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The purpose of the research is to determine if the system design for the 1980's results in isolated pairs of users. The approach is to examine the technical characteristics of digital voice links, to consider the system plans of the Services and DoD Agencies, to evaluate the mode of interoperability at the boundaries which join the several regions of the overall DoD system, and to analyze the signal-to-noise ratio (SNIR) performance of tandem-link circuits within the U.S. Army system. The study concludes that some isolated pairs of users will exist in the transitional system, but the occurrence of isolated pairs of users may be minimized by providing digital transmission paths to directly interconnect communities of 16 kB/s subscribers. The principal recommendation of the study is that tandeming of voice links should be minimized.

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DIGITAL VOICE - A KEY TO TACTICAL
COMMICATIONS IN THE 1980's

A thesis presented to the Faculty of the U.S. Army
Command and General Staff College in partial
fulfillment of the requirements of the
degree

MASTER OF MILITARY ART AND SCIENCE

by

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The opinions and conclusions expressed herein are those of the individual student author and do not necessarily represent the views of either the U.S. Army Command and General Staff College or any other governmental agency. (References to this study should include the foregoing statement.)

ABSTRACT

This research is a study of the performance of U.S. Army voice circuits in the transitional DoD telecommunications system of the 1980's. Transmission, switching, and user terminal facilities will be converted from analog to digital operation during the next two decades. Voice circuits during the transition will be composed of various tandem (series) connections of pulse code modulation (PCM), continuously variable slope delta (CVSD) and vocoder links.

The purpose of the research is to determine if the system design for the 1980's results in isolated pairs of users. The approach is to examine the technical characteristics of digital voice links, to consider the system plans of the Services and DoD Agencies, to evaluate the mode of interoperability at the boundaries which join the several regions of the overall DoD system, and to analyze the signal-to-noise ratio (SNR) performance of tandem-link circuits within the U.S. Army system.

The study concludes that some isolated pairs of users will exist in the transitional system, but the occurrence of isolated pairs of users may be minimized by providing digital transmission paths to directly inter-connect communities of 16 kB/s subscribers. The system designs for the U.S. Army System and the overall DoD System include workable interoper-ability solutions. By managing the transition in an enlightened manner, the reliability of secure voice communications may be enhanced and the operational capability to satisfy new data communications needlines may be increased.

The principal recommendation of the study is that the occurrence of tandeming of voice links should be minimized.

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Carol, my wife, cheerfully shared the life of a graduate student, again.

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TABLE OF CONTENTS

LIST OF FIGURES

LIST OF TABLES

CHAPTER 1

ANALOG AND DIGITAL VOICE COMMUNICATION

The research reported in this thesis is directed toward the broad goal of improving U.S. Army voice communications systems. DoD communications systems are beginning an evolutionary transition from predominately analog systems to predominately digital systems. The goals include economy, widespread security, and expanded capability to meet the growing data communications needs. The transition is conceived as evolutionary change over a span of two decades. But the transition involves revolutionary change in system design, doctrine, and operational concepts.

The employment of communications circuits that are tandem connections of digital voice links (i.e., series links) is essential to maintain continuity of user services during the transition. But the engineering analytical and experimental basis for design of a network with flexible use of tandem links in various circuit paths is incomplete. The tandem links in a "circuit chain" during the 1980's include analog links, and the following digital links which employ digital voice converters: (1) pulse code modulation, (2) delta modulation, and (3) vocoder links. (The vocoder, voice-coder, is the common name for digital voice converters that gain transmission economy by decomposing speech into excitation and vocal-tract components.)

In Chapter 1 we consider some technical aspects of voice communications to establish the basis for the system level problem

of tandem links. The nature of the speech signal is discussed and methods of analog and digital transmission are surveyed.

In Chapter 2 we review the importance of secure voice service as a capability of DoD systems and summarize the planning-context and tandem-link aspects of the transition to a digital system. A comparison of the performance of digital and analog voice links leads to an expanded statement of the key problem that is the focus of this research. The purpose of this research is to determine if the U.S. Army tactical communications system design for the 1980's includes isolated subscribers. The focus of the study is the performance of voice circuits of digital voice links connected in series.

In Chapter 3 we examine the digital communications system plans of the U.S. Army and other mutually-dependent segments of the total DoD system. The discussion includes an examination of the TRI-TAC Architecture, the Integrated Tactical Communications Systems (INTACS) Study, and other systems such as the World Wide Military Command and Control System (WWMCCS), the Defense Communications System (DCS), and the U.S. Navy System.

An analysis of interoperability between systems in the DoD leads to the broad conclusion that the overall system is converging toward a common target. No serious limitation of operational capability is inherent in the system level designs. Suitable interoperability solutions are being planned.

In Chapter 4 we examine a model of the 1980's transitional system in more detail, to include tandem links internal to the U.S. Army portion of the system. The available experimental and analytical results which describe the performance of digital voice links are surveyed. An analysis is performed to estimate the signal to noise ratio (SNR) performance of

tactical circuits of up to five tandem links.

In Chapter 5 the conclusions and recommendations of the study are given. The study concludes that some isolated pairs of users will exist in the transitional system of the 1980's, but the occurrence of isolated pairs of subscribers may be minimized by providing digital transmission paths to directly interconnect communities of 16 kB/s users. The system designs for the U.S. Army System and the overall DoD System include workable interoperability solutions. By managing the transition in an enlightened manner, the reliability of secure voice communications may be enhanced and the operational capability to satisfy new data communications needlines may be increased. Chapter 5 includes a brief summary of the study.

PROTECTIVE MARKING

The "For Official Use Only" protective marking on the pages of this thesis is applied due to the protective markings on six of the references cited. In particular the Bibliography entries numbered 5, 8, 20, 26, 27, and 28 possess protective markings. No protective marking is applied to the Abstract on page iii. (The Abstract is approved for public release; distribution unlimited.)

INTRODUCTION

In this initial chapter we examine the technical characteristics of various analog and digital voice links that will be employed in future systems. The nature of the voice signal is addressed in the first section by examining speech production, a simple model, and spectrum analysis. Analog transmission is the topic of the second section.

The third and most important section reports an examination of digital transmission of speech. Pulse code modulation (PCM), delta modulation (DM), and the vocoder are considered. The chapter concludes with a summary.

NATURE OF THE VOICE SIGNAL 1

The purpose of a voice communication system is to convey the voice signal from subscriber-to-subscriber in such a manner that the listener may understand the spoken words and perceive the subjective quality of the talker's speech.[2] The thrust of voice communications system planning is to devise and implement efficient methods to convey an intelligible representation of the voice signal. For these reasons, our study begins with an examination of the nature of the voice signal.

Speech Production 3

The human voice signal is an acoustic waveform produced by the interaction of the articulation of the vocal tract with the excitation of rushing air from the lungs. We examine the distinct functions of excitation and vocal-tract articulation to seek insight into the nature of the voice signal.

Excitation. Human speech originates in the larynx -- a box-like structure of cartilage at the upper end of the trachea. The larynxx

houses two lips of ligament and muscle called the vocal cords. The opening between the vocal cords is called the glottis.

Voiced speech is produced by forcing air through the glottis while the vocal cords are held under tension. The glottis vibrates open and shut generating a quasi-periodic flow of air -- a pulse train acoustic time function rich in harmonics. The fundamental frequency of the vocal cord oscillation is called the voice pitch.

Unvoiced speech is produced by a turbulent flow of air past a constriction in the vocal tract, or by a release of pressure at some point of closure in the tract. Unvoiced excitation is an acoustic noise source, a signal with a broad spectrum.

The "a" in the word "father" is an example of voiced speech. The "s" in the word "see" is unvoiced. Several sounds are generated with a combination of both voiced and unvoiced excitation; an example is the "z" in "zoo."

A key observation about the excitation signal is that it is broad-band. That is, the signal has energy spread over a wide interval of frequency. The Fourier amplitude spectrum of the excitation signal would reveal a spread of energy from less than 100 Hz to more than 10 kHz.

The broad-band excitation signal may be compared to the carrier signal in radio communication. In radio, the message is modulated onto the carrier. In human speech, the message is imposed upon the excitation signal by the articulation of the vocal tract.

Vocal tract. The message the talker wants to convey is imposed on the excitation signal by the changes in position of the tongue, lips, and other moving parts of the tract. These moving parts are called articulators and their activity in creating the spoken language is called articulation. During articulation the vocal cavity assumes different

positions causing resonances in the tract which alter the spectrum of the excitation signal. Thus, the differences between sounds perceived by a human listener depend upon both the excitation and vocal-tract articulation of the talker. The different sounds of speech are called phonemes.

Phonemes. The phoneme is the smallest unit of speech that distinguishes one utterance from another. General American English has about 42 phonemes. We may think of the set of phonemes as a code, uniquely related to the articulatory gestures of the language.

The vowel sounds of speech are produced by voiced excitation of the vocal tract (e.g., the "a" in "father"). In normal articulation the tract is held in a relatively stable position during the production of a vowel sound. Vowels usually have a duration of 60 to 200 msec. The vowel phonemes may be uttered as sustained sounds, requiring no articulatory motion.

Aside from the vowels, the remaining phonemes are referred to as consonants, some of which are more transitory in nature. Fricative consonants are produced by noise-like excitation of the vocal tract caused by turbulent air flow at a constriction. The vocal-cord source may operate in conjunction with the noise source to produce a voiced fricative (e.g., the "z" in "zoo").

Stop consonants are produced by the abrupt release of pressure at a place of closure in the tract (e.g., the "t" in "to"). The articulatory movements which generate stop consonants are more rapid than for other sounds. Stops may be voiced or unvoiced.

The remaining consonants are classified as nasals, glides, semi-vowels, diphthongs, and affricates. Further insight into the nature of vowels and consonants may be gained by examining typical waveforms.

Typical waveforms. The waveforms of two phonemes (spoken by a female talker) are shown in Figure 1. Each plot illustrates the waveform of a 51.2 msec segment of conversational speech. The scale on the abscissa is 12 msec/inch, and scale marks are spaced at 5 msec intervals. The stop consonant "t" in "Gif<u>t</u> is" is shown in Part (a). The closure portion of the "t" lasts about 40 msec. The silence of closure is followed by a burst, signaling the release of pressure at the closure. The burst response decays with roughly exponential envelope to a low level aspirated noise. (To aspirate is to pronounce with a breathing, as in the fricative "h" in "<u>h</u>e".) The burst decays in 10 msec, while the following aspiration lasts about 20 msec before the buildup of the vowel "i" in "gift <u>i</u>s". A 51.2 msec segment of the vowel "i" is shown in Part (b). The last 5 msec of the aspirated noise is displayed at the extreme left. Voicing begins with a pitch of approximately 185 Hz, and within about 15 msec the waveform displays the stationary character of a sustained vowel. The maximum amplitude of the vowel in Part (b) is five times that of the stop consonant in Part (a). [4]

Notice the noise-like nature of the unvoiced stop consonant in Part (a). In contrast, the vowel in Part (b) is stationary over an interval, and is almost periodic. The periodic segments repeat at 5.4 msec intervals. (The reciprocal of the 5.4 msec pulse interval yields the pitch frequency, 185 Hz.)

In summary, we conclude that excitation and vocal tract articulation are the two distinct functions in the production of human speech. This simple view of speech production motivates a model that is widely used in analysis - synthesis devices which communicate the human voice signal in an efficient manner.
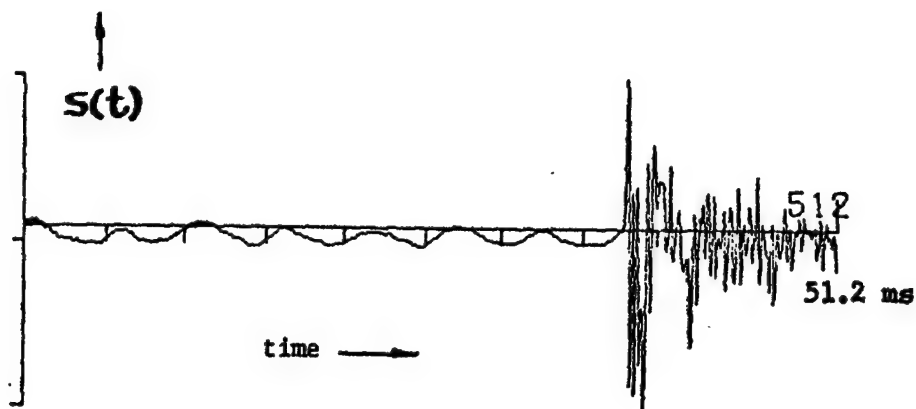
(a)
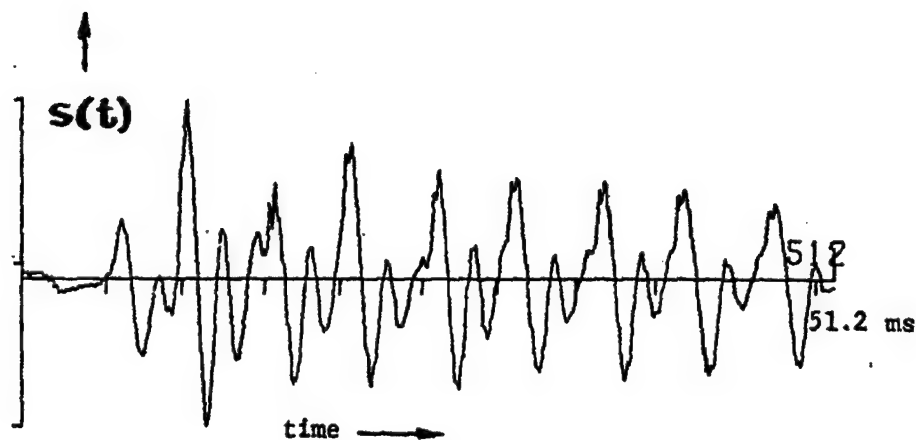
(b)

Figure 1. Typical Speech Waveforms (Female Talker)

(a) The Stop Consonant "t" in "gift is,

(b) The Vowel "1" in "gift is"

The Simplified Model

The speech waveform may be decomposed into excitation and vocal tract components. Such decomposition is the central strategy of nearly every speech analysis-synthesis system. The strategy was conceived by Homer Dudley at the Bell Telephone Laboratories and incorporated into his invention of the channel vocoder in 1939.[5] Speech bandwidth reduction (or data rate reduction) research during the last three decades has been dominated by the intuitive strategy pioneered by Dudley.

The speech signal may be viewed as a response of a linear system (the vocal tract "filter") to one or more sound sources (i.e., the excitation). This statement is the essence of the acoustic theory of speech production described by Gunnar Fant.[6] During a voiced sound, a periodic pulse-train excitation signal $e(t)$ is the input to the linear time-invariant vocal-tract filter which has impulse response $v(t)$. The resulting speech waveform $s(t)$ is the convolution of $e(t)$ with $v(t)$.[7]

$$s(t) = e(t) \otimes v(t) = \int_{-\infty}^{\infty} e(\gamma)\, v(t-\gamma)\, d\gamma \qquad (1-1)$$

In this simplified model, unvoiced sounds are assumed to be produced in the same fashion, except that $e(t)$ is considered to have the character of stationary random-noise.

The word "stationary" applies to the three assumptions of the model: (1) $v(t)$ is the impulse response of a time-invariant filter (i.e., the articulators are not moving); (2) For unvoiced sounds, $e(t)$ is considered to be a stationary random process; (3) For voiced sounds, $e(t)$ is periodic.

The articulatory gestures in speech production are relatively

slow -- compared to a pitch period -- so we may think of the speech waveform as being constructed of short segments, each segment corresponding to a fixed vocal-tract configuration. The motion of the articulators is continuous, so the simplified model may be used for successive short segments during which only incremental movement of the tract occurs.

The simplified model of speech production is illustrated in Figure 2. Typical waveforms for $e(t)$, $v(t)$, and $s(t)$ for voice speech are sketched at top of Figure 2. The corresponding amplitude spectra $|E(f)|$, $|V(f)|$, and $|S(f)|$ are sketched at the bottom of Figure 2. [8] Notice that the interval between pulses in $e(t)$ is the pitch period, τ_0. [9] The lines in the spectrum $E(f)$ are spaced $1/\tau_0$, representing the fundamental pitch frequency and the set of harmonic frequencies. [10]

Decomposition of speech into excitation and vocal tract components using the simplified model is the key feature of that class of analysis-synthesis systems known as vocoders. We return to the subject of analysis-synthesis in a subsequent section.

Another description of the strategy of the vocoder is that it attempts to preserve the short-time spectrum of the speech signal. We examine the short-time spectrum in the next section.

Spectrum Analysis of Speech

The traditional tool for spectrum analysis of signals and linear systems is the Fourier transform pair.

$$S(f) = \int_{-\infty}^{\infty} s(t) e^{-j2\pi ft} dt \quad , \quad s(t) = \int_{-\infty}^{\infty} S(f) e^{+j2\pi ft} df \quad (1-2)$$

The speech signal, however, is not known over all time. In addition, delay in excess of several hundred milliseconds is not tolerable
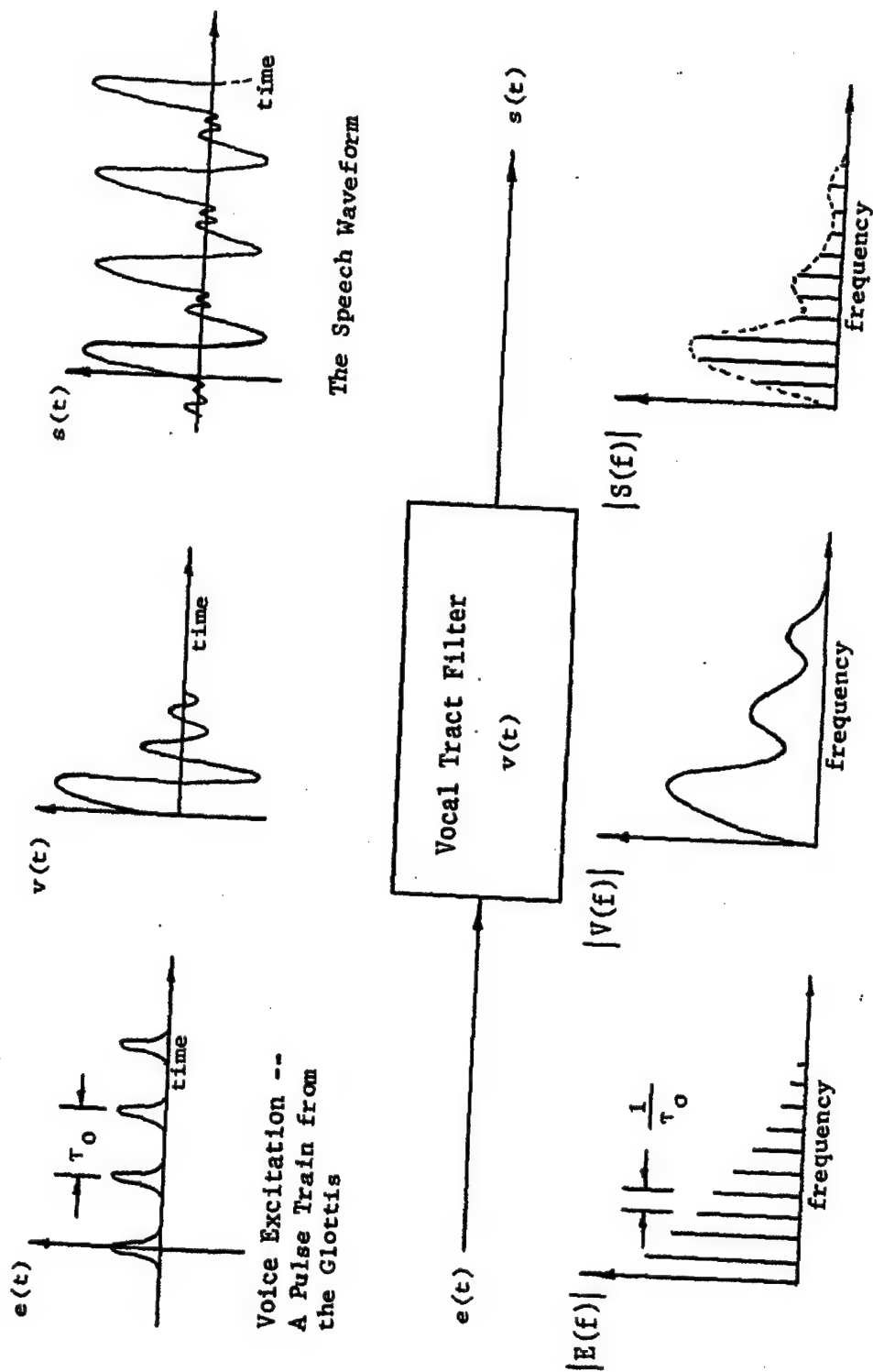
Figure 2. The Simplified Model of Speech Production

in voice communications. Furthermore, the stationary assumptions of the simplified model are only valid over short segments of the speech signal. For these reasons, the Fourier transform of equation (1-2) must be modified.

Short-time spectrum. For speech we desire a frequency representation which puts in evidence the spectral content of short segments of the waveform. To this end, we choose a window function $w(t)$ which is causal (i.e., $w(t) = 0$ for $t < 0$) and essentially non-zero only over a duration D. Forming the product

$$s(\tau) \, w(t-\tau)$$

and Fourier transforming yields the short-time spectrum $S(t,f)$.

$$S(t,f) = \int_{t-D}^{t} s(\tau) w(t-\tau) e^{-j2\pi f\tau} d\tau \qquad (1-3)$$

The short-time spectrum $S(t,f)$ is the Fourier transform of the recent past of the time function $s(\tau)$ weighted by the window function $w(t-\tau)$.[11] The short-time spectrum is the key analysis tool employed in various implementations of the vocoder. $S(t,f)$, measured by analysis at successive time intervals $t = t_1, t_2, \ldots$, is smoothed in frequency to remove the influence of the pitch harmonics. The smoothed version of the short-time amplitude spectrum $S(t_1, f)$ is an approximate measurement of the vocal-tract amplitude spectrum for the time epoch corresponding to t_i. (Smoothing may be accomplished by convolution, e.g., time functions are smoothed by low-pass filtering.)

Digital spectrum analysis. Digital spectrum-analysis is accomplished with the discrete Fourier transform (DFT) pair:

$$S(KF) = \sum_{n=0}^{N-1} s(nT) e^{-j2\pi nK/N}$$

$$(1-4)$$

$$s(nT) = \frac{1}{N} \sum_{K=0}^{N-1} S(KF) e^{+j2\pi nK/N}$$

T is the sampling interval of the time function s(t); N is the number
of samples to be transformed; and F = 1/NT is the sampling of the spec-
trum. [12]

To obtain the discrete short-time Fourier transform, we intro-
duce the window function into equation (1-4):

$$S_n(KF) = \sum_{n=0}^{N-1} w(nT)s(nT + rMT) e^{-j2\pi nK/N} \tag{1-5}$$

The index r corresponds to the running time variable in S(t,f). The
short-time spectrum is evaluated at times t=r MT for r=1,2,..... The
window is propogated along the time function in steps of MT seconds.

The ease of implementation of the DFT on modern digital com-
puters has contributed to the increased interest in digital voice
research in the last decade. Short-time spectrum analysis is a very
powerful tool in studying the nature of speech and in representing
speech efficiently.

Human hearing. There is evidence that the ear performs a mech-
anical short-time spectrum analysis at an early stage of processing.
Flanagan has advanced a filter model for this observed effect. [13]

In the middle of the last century, Ohm formulated his famous
law which stated that "aural perception depends only on the amplitude
spectrum of a sound and is independent of the phase angles of the var-
ious frequency components contained in the spectrum." [14] The contro-
versy over the ability of the human to perceive phase continues. It

is widely accepted, however, that the human hearing process is predominantly insensitive to phase.

Moderate phase distortion in a telephone channel, for example, does not result in perceptable degradation of voice quality to the listener. In the context of vocoder communications, phase information is discarded to achieve economy of transmission (i.e., only the amplitude spectrum is conveyed to the receiver).

Summary

The physical functions of speech production are excitation and vocal-tract articulation. Excitation is voiced and quasi-periodic, or unvoiced and noise-like. The excitation signal spectrum is broad in frequency. The message is imposed on the excitation signal by the vocal tract. The phonemes of speech differ in one or more features: (1) the state of voicing, (2) the duration, and (3) the gross shape of the short-time spectrum.

The simplified model of speech production is a stationary representation which describes speech as the response of a linear filter. The use of short-time spectrum analysis is the method used in the vocoder to measure the effect of the vocal tract in successive segments of speech.

ANALOG TRANSMISSION - THE TELEPHONE CHANNEL

"Transmission is the part of communications engineering concerned with transmitting messages between sources and receivers." [15] For many decades the focus of transmission engineering was the analog channel. The key concerns were defining and maintaining bandwidth, loss and interference objectives, and fielding economical multiplexers

and radio links in a reliable switched network.

The microphone in a telephone handset is a transducer that transforms the acoustic voice signal (the source) into an electrical signal — the electrical analog. The analog telephone circuit conveys the signal to the receiver (the sink-the human listener) through a "pathway" of electromagnetic potential. The pathway may be composed of many links of wire, cable, and radio. An analog transmission link retains the input signal in some form of a continuous waveform.

The Linear Channel

An analog transmission link may be modeled as a linear system with additive noise. Such a link is illustrated in Part (a) of Figure 3. The impulse response of the channel is $h(t)$. The Fourier transform of $h(t)$ is $H(f)$. The following relationships hold.

$$y(t) = s(t) \otimes h(t) + n(t)$$

$$Y(f) = S(f) H(f) + N(f) \tag{1-6}$$

The first term on the right side of (1-6) is the signal component of the output, which has magnitude $|S(f)| \cdot |H(f)|$ or converting to decibels (dB) becomes $|S(f)|_{dB} + |H(f)|_{dB}$ Thus, the linear channel modifies the input by attenuation of some frequency components, and by adding noise.

The attenuation of a typical channel (of one link) is sketched in Part (b) of Figure 3. The channel response begins to roll-off at frequencies below about 400 Hz, and cuts off sharply near 200 Hz. At the high frequency end, the response cuts off at about 3.5 kHz. Such a channel is called a "3 kHz channel." (The dotted curve in Part (b) illustrates the response of a 2.5 mile non-loaded cable channel.)
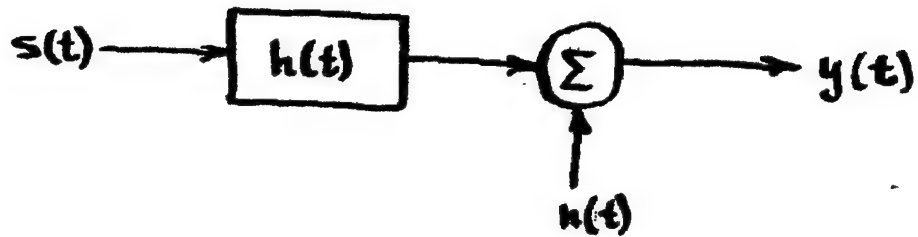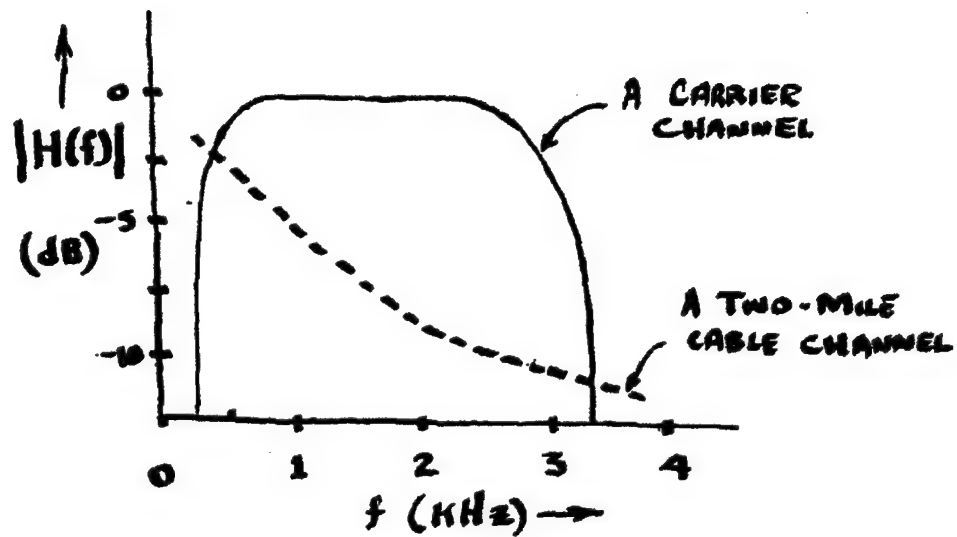
Figure 3. The Linear Channel

(a) The Channel Model, (b) Attenuation

of the Channel.

Consider a circuit that is composed of five channel links in tandem. (Tandem links are links connected in series, to establish an end-to-end circuit.) The version of the speech signal at the output reflects the cumulative attenuation of the five links, and would be accompanied by the noise contributions of each link. Thus, the topology of the telephone network must be engineered in concert with the choice of message channel objectives.

Message channel objectives in the Bell System include loss and volume, frequency response, noise, and echo. Noise includes random noise, crosstalk, low frequency hum, impulsive noise, and intermodulation (due to nonlinearities in the channel). A typical design objective for noise on a 4000 mile link in the Bell System is -46 dBm. (dBm is a measure of power relative to one milliwatt.) [16]

The linear channel model is a useful one that is reasonable to describe the performance of analog channels and "high quality" digital channels (such as the PCM used in the Bell System) for conveying voice signals. The performance of an analog channel for conveying non-voice signals (e.g., analog facsimile and digital modem signals) is limited by impairments that are not adequately described in the linear model (e.g., phase distortion and various nonlinearities).

Volume and Signal-to-Noise Ratio

A simple periodic voltage can be measured by several methods: the peak, the average, or the root-mean-square (rms). Each measure may be used to compute power. A sinusoidal test-tone "signal" of power 0 dBm if combined with a noise power of -46 dBm would yield a signal-to-noise ratio (SNR) equal to 46 dB. But speech is not a simple, periodic signal.

The nature of speech is such that the average, rms, and peak values (and the ratio of one to the other) are all irregular functions of time. A practical method of measuring speech volume has evolved, using an rms meter with an integrating time-constant of syllabic duration. Such a measurement provides a useful measure of speech power in volume units (vu) on a dB scale. In this manner, a signal volume measure of -6 dBm combined with a -46 dBm noise may be interpreted as a SNR=40 dB. (SNR measurements or "predictions" must be interpreted with caution. For example, if a talker's input volume is 20 dB below the nominal level, the SNR at the receiver may be degraded by 20 dB. Most SNR calculations and measurements are performed for some rather simple "test-signal" such as an 800 Hz sinusoid, a triangular wave, or a stationary "Gaussian signal." Such calculations and measurements are useful, but do not establish channel performance for either voice or non-voice signals.)

Summary

The analog channel is the central consideration in the design of a switched telephone network. The channel may be modeled as a linear system, with additive noise. The key impairments are loss, attenuation, noise of various sources, and echo. Signal-to-noise ratio is an important measure of performance. A typical long distance call in the Bell System would deliver an end-to-end circuit with SNR better than 35 dB.

DIGITAL TRANSMISSION OF SPEECH

Digital communications has been a field of rapid growth over the past 15 years. Many diverse factors converge to hasten the conversion of systems and subsystems to digital. Some of these factors are

(1) the economy of digital transmission for expansion of circuit capacity in urban areas, (2) the reliability of digital hardware, (3) improvements in device technology, (4) the growth of computer applications with increased need for data communication, and (5) the need for security. The last of these factors is perhaps the dominant one in military systems.

A signal may be secured against intercept and decoding only by digital encryption. [17] Thus, secure voice service requires conversion of speech to digital form and a digital communications network to interconnect subscribers. In this section we examine the mechanism of conversion of speech to digital (with the companion conversion back to analog).

Digital Signal Processing

The steps in converting an analog signal to a digital signal are sampling, quantization, and coding. We define a __digital signal__ to be a time function of binary pulses or binary "states." A digital signal may take on one of two values (or states) in each pulse interval. [18] __Sampling__ converts the analog signal to one that is discrete in time. __Quantization__ converts time discrete samples (of continuous amplitudes) into discrete amplitudes. __Coding__ converts the time and amplitude discrete samples into binary pulses. The process of analog-to-digital (A/D) conversion is then complete.

The digital signal is conveyed to the receiver (or to a point of analog interface) and the inverse steps are employed to reconstruct the analog signal. The reconstruction process is called digital-to-analog (D/A) conversion.

__Sampling.__ The first step in A/D conversion is sampling.

Sampling converts an analog signal s(t) into a "stairstep" function
which has amplitudes s(nT) in each time interval of width T, for n =
0,1,2,... . The sampling theorem for low-pass, band-limited signals
follows:

> A low-pass, band-limited function having no frequency
> components outside the interval of frequency from -F to
> F Hz may be described uniquely and completely for all time,
> by the set of sample values taken at time instants separated
> by 1/2F seconds or less. [19]

The sampling theorem provides a mathematical basis for converting an
analog signal into a time-discrete one which can be reconstructed identi-
cally to the original signal. In practice, hardware limitations make
the reconstruction less than perfect, but the imperfections may be
essentially negligible. The dominant impairment in the A/D - D/A pro-
cess is quantization.

Quantization. Linear quantization is the "rounding off" of the
amplitude of each time sample to one of a discrete set of amplitudes.
The process is essentially the same as rounding off a series of numbers
to a fixed number of decimal places. A linear quantizer is one with equal
quantum steps between levels.

Binary Coding. The final step in A/D conversion is the coding
of time and amplitude discrete samples into binary pulses. Suppose the
signal is sampled at intervals of T seconds, and the quantizer has

$$L = 2^B = 2^8 = 256$$

levels. Each of the 256 levels may be uniquely labeled (or signaled) by
a B = 8 bit binary number (or word). (A bit is a binary digit, with value
zero or one.) In this manner a signal that is sampled at intervals T
seconds and quantized to 256 levels may be converted by coding to a

digital signal of binary pulses with pulse interval T/8 seconds.

An example of practical importance is pulse code modulation (PCM). Speech is low-pass filtered to 3 kHz and sampled at $1/T = 8$ kHz. Quantization to $L = 256$ levels is followed by binary coding to $B = 8$ Bits/sample. The resulting digital signal rate is

$$I = B/T = (8 \text{ Bits/Sample}) (8000 \text{ Samples/sec})$$

The channel rate for 8 Bit PCM is 64,000 Bits/sec (64 kB/s). (The symbol I represents the bit rate of a digital channel or a digital signal.)

Pulse Code Modulation (PCM)

The most widely employed digital voice technique is pulse code modulation (PCM). PCM was developed in the 1950's and fielded in the Bell System beginning in the early 1960's. Bell Telephone T-1 facilities are deployed now in many U.S. cities, and expansion continues. [20]

Linear PCM. The steps in linear PCM, discussed above, are illustrated in the diagram of Figure 4. The staircase function is the input versus output characteristic of the quantizer. (Note that the quantizer is not a linear system. A quantizer with uniform quantum-step size is commonly referred to as a "linear" quantizer. Thus, the label linear PCM is common.) Quantization is the deliberate and dominant source of impairment in PCM. The error introduced by quantization represents a loss of information, and is called quantization noise (shown as e_n at the quantizer output). After passing through a channel (assumed to be error-free) the digital signal is decoded to an L level staircase version (equivalent to that at the quantizer output). Low-pass filtering completes the D/A process.

The following expression may be derived for one link of linear PCM: [21]
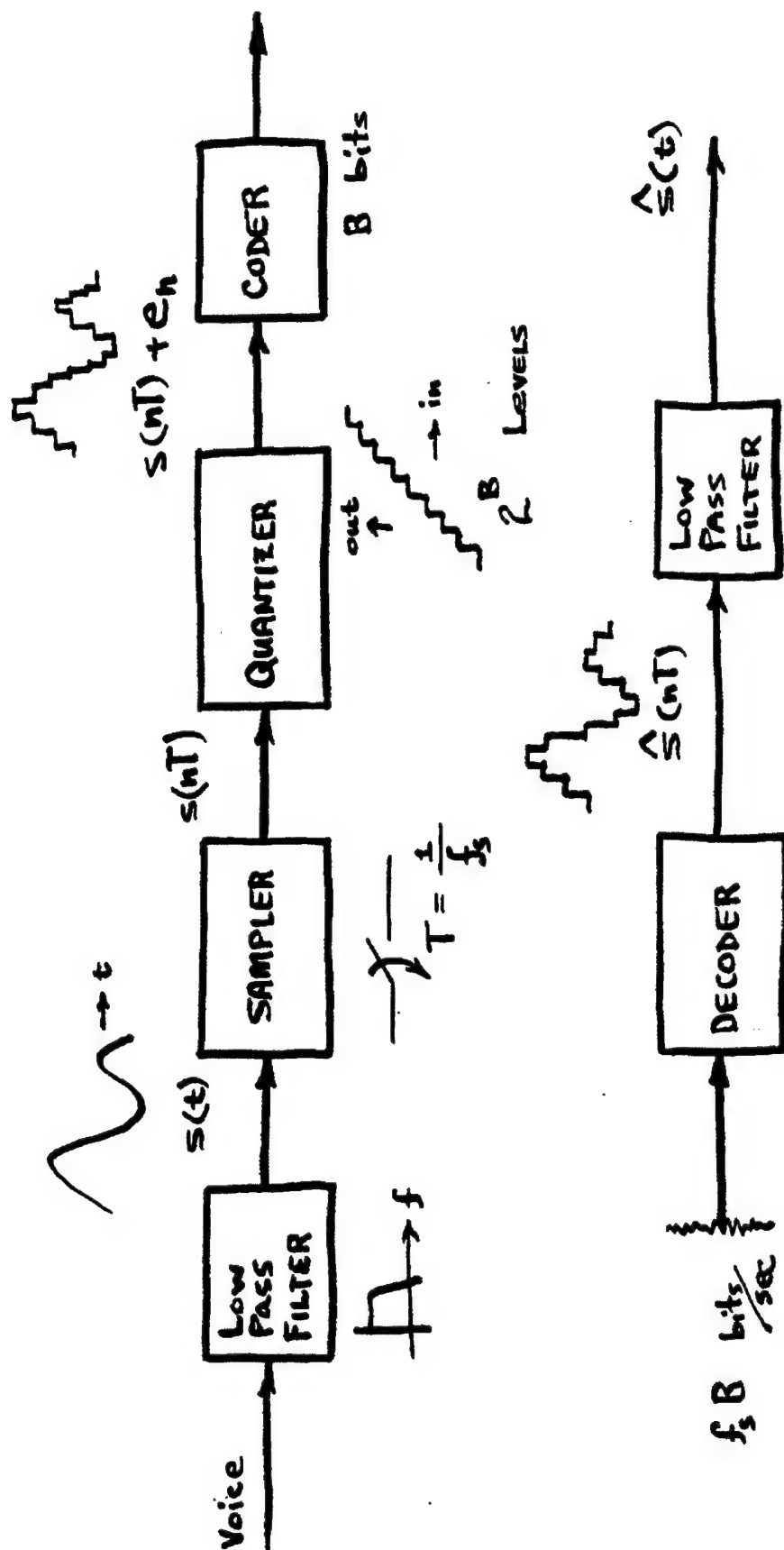
Figure 4. Linear Pulse Code Modulation

$$SNR = 6B - 7.2 \quad \text{(in dB)} \qquad (1-7)$$

Equation (1-7) reveals that one link of 8 Bit PCM should yield SNR = 40.8 dB. The equation was derived for a triangular "signal" with an input level that exactly fills the amplitude range of the quantizer. Notice that if the input signal level were increased, some samples would fall outside the quantizer range, causing overload distortion and decreasing the SNR. Similarly, decreasing the input signal below the optimum level decreases the SNR by an identical amount. This discussion highlights the key weakness of linear PCM, poor dynamic range.

The phoneme sounds of speech vary in power over a range of 10 to 15 dB, and variation among different talkers gives a spread of 10 dB or so. Thus, SNR performance is important over a range of input signal level of 20 to 30 dB. (Another important aspect is the tandeming of links, a topic we return to below.)

Companded PCM. Companding is a method of improving the dynamic range of a voice trunk. Companding means compression followed by expansion. Companding may be thought of as a nonlinear gain-loss mechanism that amplifies weak signals (at the transmitter) to improve the SNR performance on the channel (compression). Attenuation of (the previously amplified) weak signals at the receiver (expansion) restores the signal to its original (amplitude) balance.

Instantaneous companding is employed with PCM using a nonlinear (logarithmic) compression characteristic. The companding is performed on a sample-by-sample basis, with no interaction between samples (i.e., instantaneous companding, the process has no memory). In the U. S., Japan and several other countries the compression law employed is the μ-Law

curve:

$$y = \frac{V \ln(1 + \mu x/V)}{\ln(1 + \mu)} \qquad (1-8)$$
$$0 \leq x \leq V$$

Where X is the compressor input, Y the output, V the clipping level (the maximum range of the quantizer), and μ is the compression design parameter. The curve is odd symmetric for negative inputs. [22]

The SNR performance for companded PCM is illustrated in Figure 5 as a function of input signal level (ISL). ISL is given in dB below the clipping level.

Curve (a) shows the rugged SNR performance of the 6-bit/sec PCM used in the Bell System, SNR > 36dB over a 40 dB range of ISL. By contrast curve (b) shows the SNR performance of the 48 KBit/sec companded PCM used in the U.S. Army TD-660 multiplexer. [23] (Notice that on either side of the twin peaks of the curve, SNR falls off at the rate of about .8 dB per dB change of ISL. This property appears to be important in the performance of tandem links.) Curve (c) is an approximation of the SNR characteristic for CVSD, a version of delta modulation.

Delta Modulation (DM) [24]

Delta Modulation (DM) is a digital voice technique that employs one-Bit quantization of the difference between successive samples. The binary difference is a measure of slope, coded as positive or negative (i.e., binary -1 or 1). An integrator feedback path forms a local reconstruction of the prediction of the signal (at the transmitter). DM delivers improved performance over PCM at channel rates below about 30 KBits/sec, and is a very robust performer over a digital channel of high error rate.

Linear DM suffers the same limitation as linear PCM, i.e., poor
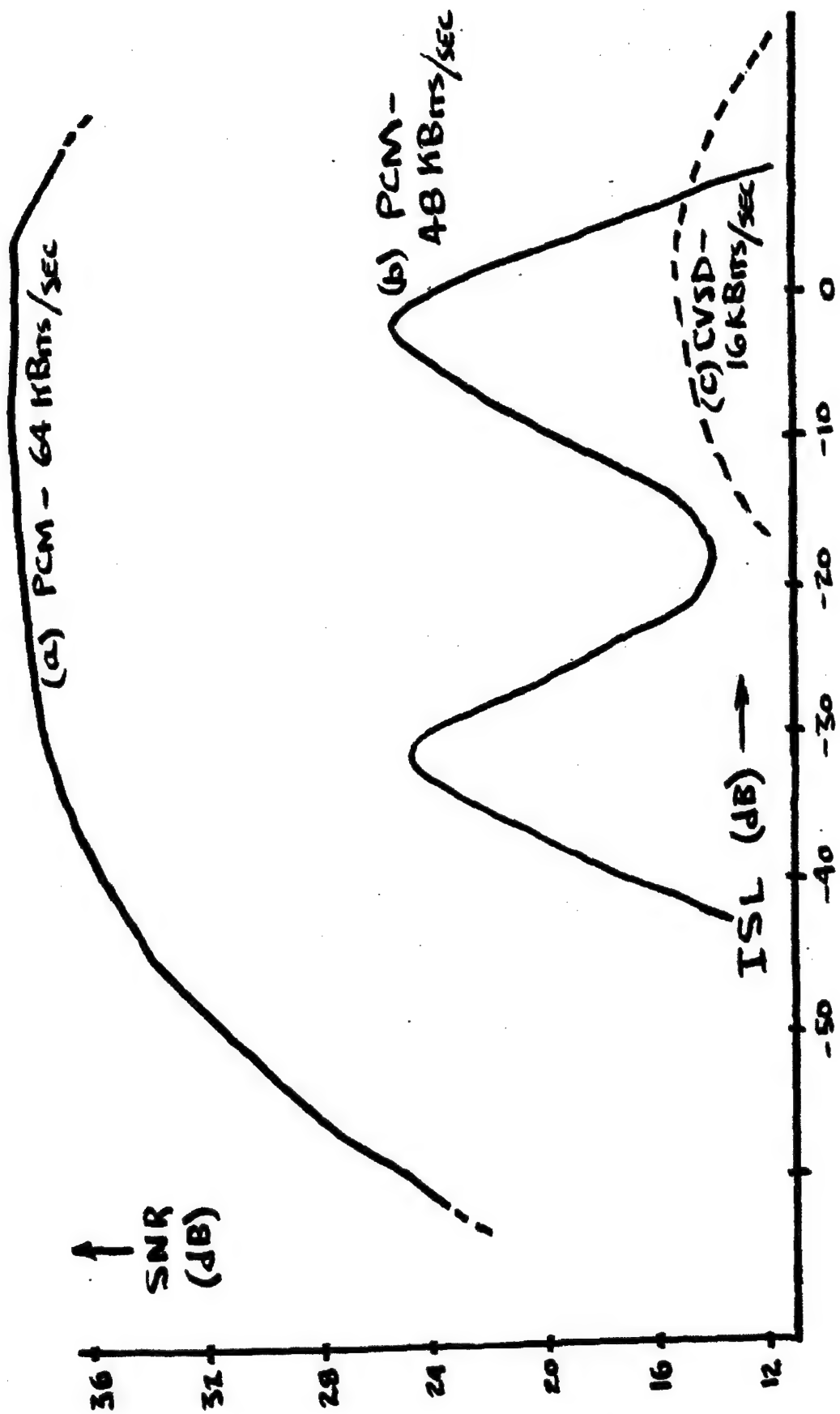
Figure 5, Digital Voice SNR Performance
(a) 64KB/s PCM (8 Bit), (b) 48 KB/s PCM (6 Bit)
(c) 16 KB/s CVSD

dynamic range. To improve the dynamic range performance various forms
of companding have been implemented. We focus here on the version se-
lected for use in the Department of Defense, i.e., continuously variable
slope delta (CVSD) modulation. CVSD employs adaptive companding that ad-
justs the effective slope signaled by each binary output digit. The
adjustment is controlled by the weighted average of slope-overload con-
ditions, summed over an interval of syllabic duration. The effect of the
companding is to vary the step-size of the binary slope quantizer to fol-
low the short-time energy fluctuations in the signal.

Curve (c) of Figure 5 illustrates the SNR performance of CVSD
with channel rate 16 KBits/sec. The peak SNR is about 16 dB. The width
of the flat region of near maximum SNR performance depends on the choice
of compander design parameters. (In particular, the choice of the com-
panding ratio, maximum slope to minimum slope, dominates the dynamic range
of CVSD.)

PCM and DM are digital voice techniques that seek to reconstruct
a replica of the original signal at the receiver that is corrupted only
by quantizing noise. A different strategy is employed in analysis-synthesis
techniques.

Analysis-Synthesis, the Vocoder [25]

The strategy of analysis-synthesis techniques such as the vocoder
("voice-coder") is to achieve low digital channel rates by decomposing
speech into excitation and vocal-tract components. Excitation may be ef-
ficiently coded into one number (5 to 7 Bits) for each stationary interval
(or analysis-frame). Excitation is coded to signal the state of voicing
(i.e., voiced or unvoiced) and an estimate of pitch frequency (for voiced
sounds).

The vocal-tract description is analyzed by short-time spectrum analysis at the transmitter. A quantized approximation of the "vocal-tract filter" is conveyed to the receiver. Reconstruction of a synthetic version of speech is accomplished at the receiver by exciting a time-varying filter which employs the received vocal-tract parameters with a pulse or noise excitation.

The vocoder process is identical to the simplified model of speech production. The process is performed in sequence for each "frame" of input speech. A typical frame interval is 20 msec or 30 msec.

The channel vocoder has been employed in military systems for many years. A channel vocoder is one that employs a bank of channel filters to measure the approximate vocal-tract spectrum. A typical channel rate is 2.4 KBits/sec.

The key advantage of the vocoder is its low channel rate. (A typical vocoder channel rate is I = 2.4 KB/s, a rate low enough to be conveyed over an analog telephone channel by employing a modem to convert the digital signal to a "quasi-analog" one.) The disadvantages include: (1) cost and complexity (e.g., more than $10,000 per terminal, several cubic feet in volume, several hundred pounds in weight, several hundred watts of power required), and (2) relatively poor voice quality. Vocoder speech displays a "machine-like" quality that obscures the perception of the talker's identity and of the emotional nuances of speech. The vocoder does provide speech of useful intelligibility ($\approx 90\%$). User acceptance of vocoder quality speech is difficult to gain.

Significant progress is now being made with the technology of linear predictive coding (LPC). LPC vocoders employ the same strategy as the channel vocoder (i.e., efficient coding of the voice signal by decomposition into excitation and vocal-tract components), but use the method

of linear prediction to measure and code a set of parameters to describe the vocal-tract spectrum. LPC offers key improvements in hardware simplicity and potential improvements in voice quality.

Vocoders remain limited in voice quality and naturalness due to the approximations of the simplified model employed and the practical difficulties in measuring pitch and voicing.

SNR is not a useful measure of performance for vocoded speech because the vocoder does not attempt to reconstruct the original waveform. (Instead, the approach is to construct a synthetic version which has approximately the same short-time amplitude spectrum as the original.) The widely used measure of goodness for vocoded speech is word intelligibility. Subjective quality listening tests aid in vocoder comparison.

Vocoded speech may not be usefully modeled as original signal plus noise. The removal of redundancy accomplished by the vocoder to gain low channel rates is counter-balanced by increased vulnerability due to various nonlinear effects. The vocoder does not provide an "analog channel" that may readily operate in a tandem link connection. For example, if the speech at the input of a vocoder A/D is already degraded in bandwidth and by noise, then the ability of the vocoder analyzer to measure excitation and vocal tract parameters may be impaired. [26] Thus, the ability of the vocoder to perform in tandem link circuits is a key determinant of network configuration and operational capability.

Errors on the digital channel contribute noise in the D/A reconstruction, and in severe error situations may result in receiver instability.

Digital transmission of speech is summarized in the chapter summary which follows.

SUMMARY

The nature of the voice signal is important to any study of voice communications performance or digital voice techniques. Speech is produced by vocal-tract articulation imposed on the excitation signal from the glottis. The simplified model employs stationary assumptions. Short-time spectrum analysis is useful to implement the stationary model in the vocoder.

The analog telephone channel is an approximately linear system with various impairments (loss, attenuation, noise, and echo). SNR is a key performance measure.

Digital voice communications involves A/D conversion, transmission in a digital channel, and reconstruction by D/A conversion. The A/D and D/A process causes reconstruction impairment due to quantization noise and nonlinear corruption of the voice signal. Impairments due to errors on the digital channel include noise and potential receiver instability.

The steps in the A/D - D/A process include the digital signal processing techniques of sampling, quantization, and binary coding. Companding is commonly employed to improve the dynamic range of digital voice converters.

Pulse code modulation is a widely used technique that delivers excellent SNR performance with $B = 8$ Bit coding, and channel rate $I = 64$ KBits/sec. Instantaneous companding with the μ-Law nonlinearity provides a wide dynamic range. The $B = 6$ Bit PCM ($I = 48$ KBits/sec) used in U.S. Army systems displays a "twin-peak" SNR versus ISL characteristic because of the simple compander characteristic.

Delta modulation is a relatively simple digital voice technique that delivers channel error and SNR advantages over PCM at channel rates in the range $I = 10$ to 30 KB/s. Continuously variable slope delta (CVSD)

modulation is a version of DM that employs adaptive slope companding. CVSD performance gains over PCM result from matching the A/D conversion process to the nature of the speech signal and to subjective perception of the human listener. (A channel of 16 or 32 KB/s CVSD is inferior to 64 KB/s PCM or analog channels in ability to convey non-voice signals such as modem signals.)

The vocoder achieves low channel rates by decomposing speech into excitation and vocal-tract parameters. LPC vocoders may gain performance improvements.

The technical examination of digital voice techniques motivate several questions: (1) What ability do these techniques have to deliver satisfactory performance if operated in tandem link connections? (2) Does the ability to tandem imply any constraint on network design, or impose any limitations of operational capability? We examine the important question of tandem link performance in Chapter 2.

CHAPTER END NOTES

1. The basic reference for this section is James L. Flanagan, Speech Analysis Synthesis and Perception (New York: Springer-Verlag, 1972), pp. 9-139. Portions of this section are adapted from Jack C. Hammett, Jr., "An Adaptive Spectrum Analysis Vocoder" (unpublished Ph.D. dissertation, Georgia Institute of Technology, 1971), pp. 5-36.

2. Subjective quality includes properties of a voice communication circuit that are not readily quantified. The ability of the listener to recognize the identity of the talker is one aspect of subjective quality. Another aspect is the ability of the listener to perceive the emotional content conveyed in the talker's voice.

3. Flanagan, pp. 9-22.

4. Hammett, pp. 9-11.

5. Homer Dudley, "The Carrier Nature of Speech," The Bell System Technical Journal, vol. 19 (October, 1940), pp. 495-515.

6. Gunnar Fant, Acoustic Theory of Speech Production (The Hague, Netherlands: Mouton and Co., 1960).

7. George R. Cooper and Clare D. McGillem, Methods of Signal and System Analysis (New York: Holt, Rinehart, and Winston, Inc., 1967), p. 120.

8. The symbols $|\cdot|$ used in $|E(f)|$ indicates the absolute value (i.e., magnitude) of the function $E(f)$. $E(f)$ is a fourier transform (i.e., spectrum), a complex function of frequency, so the amplitude spectrum

$|E(f)|$ is the square root of the sum of the squares of the real and imaginary parts of $E(f)$. Taking the magnitude of a function discards phase information.

9. The symbol "γ" in equation (1-1) is a dummy (time) variable of integration. The symbol γ_0 in Figure 2 is a number -- the pitch period.

10. Hammett, pp. 11-15.

11. Flanagan, pp. 141-155.

12. Bernard Gold and Charles M. Rader, Digital Processing of Signals (New York: McGraw-Hill Book Co., Inc., 1969), pp. 3-18.

13. Flanagan, pp. 108-126.

14. Manfred R. Schroeder, "Models of Hearing," Proceedings of the IEEE, vol. 63 (September 1975), pp. 1332-50.

15. Bell Telephone Laboratories, Transmission Systems for Communications (Winston-Salem: Bell Telephone Laboratories, Inc., 1964), p.v.

16. Bell Telephone Laboratories, pp. 16-48.

17. A satisfactory method of analog encryption to gain security has not been invented. A secure signal is one that is encrypted. An encrypted signal may be intercepted by an unauthorized party but may be decoded to obtain the information signal only by transforming the encrypted signal with the proper decryption algorithm.

18. The most simple version of a digital signal is a rectangular

waveform which takes on one of two values in each pulse interval, e.g., zero or one volt. If the binary pulses modulate a carrier waveform (in phase, frequency, or amplitude) for transmission by radio or cable, the transmitted version (although "analog" in some respects) is a digital signal (because the information signal is binary, and the transmitted version takes on only one of two "states" in each pulse interval.) Multiplexing several digital signals into an interleaved stream results in a digital transmission signal (of increased binary rate.) Similarly, by encoding two information pulses into one signaling interval (of four states) results in a digital signal (each information pulse is signaled by one of two distinct states per pulse.)

19. Cooper and McGillem, pp. 131-3.

20. Bell Telephone T-1 facilities are deployed now in more than 45 U.S. cities, providing more than 32 million voice channel miles. Growth continues at a rate of 10,000 channel miles per day.

21. Seymour Stein and J. Jay Jones, Modern Communication Principles (New York: McGraw-Hill Book Co. Inc., 1967), pp. 205-9.

22. Bernard Smith, "Instantaneous Companding of Quantized Signals," Bell System Technical Journal, vol. 55 (May 1957), pp. 653-709.

23. The performance of 48 kB/s PCM is examined in more detail in Chapter 4.

24. N. S. Jayant, "Digital Coding of Speech Waveforms: PCM, DPCM, and DM Quantizers," Proceedings of the IEEE, vol. 62 (May 1974), pp. 611-32.

25. Flanagan, pp. 323-33.

86. One example of degraded vocoder performance is the case of sinusoidal noise at the input to the vocoder A/D. Acoustic background noise (e.g., the airborne application) or induced hum (100 Hz noise is common in communications facilities) present at the input to the A/D may degrade the pitch tracking function. (If of sufficient amplitude, the noise is "tracked" rather than voice pitch.) A related example is that if the input signal is band limited (by an analog carrier channel) or has suffered attenuation (as on a wire loop) then the vocoder vocal-tract measurement function may be degraded (because the signal spectrum to be measured is partially obscured.) Refer to Figure 3 for examples of band-limited or attenuated channels and loops.

CHAPTER 2

THE PROBLEM - SECURE VOICE, KEY TO THE TRANSITION
OF TACTICAL COMMUNICATIONS SYSTEMS TO DIGITAL
OPERATION IN THE 1980'S

In this chapter we consider the importance of secure voice, the
transition to digital operation, a comparison of analog and digital voice
links, and examine digital voice tandeming.

IMPORTANCE OF SECURE VOICE

A historic deficiency in DoD communications systems is the lack
of widely available secure voice service. Mr. Thomas C. Reed, the Direc-
tor, Telecommunications and Command Control Systems, Office of the Secre-
tary of Defense, recently summarized the situation as follows:

> Lack of security in voice communications used in the command and
> control of forces has long been recognized as contributing to reduced
> force effectiveness by allowing the enemy to gain advance knowledge
> of planned actions. Recent gains in methods of voice digitization
> and the use of large-scale integrated circuits to reduce costs and
> size now bring forth the promise of their widespread use in the field.[1]

The experience gained in recent warfare confirms the need to pro-
tect our voice communications from exploitation.[2] Because secure voice
requires digital transmission, the need for expanded secure voice ser-
vice is one key driving force in the transition of DoD systems to digital
operation.[3]

TRANSITION TO DIGITAL OPERATION

U. S. military telecommunications systems are now beginning a
conversion from analog methods to digital technology. Digital systems

convey human speech by encoding the voice waveform in binary digits for transmission. Digital signals may be readily encrypted to deny hostile exploitation. The thrust of DoD policy is that future communications systems will be digital.

The TRI-TAC Program

The Joint Tactical Communications Program (TRI-TAC) was established by the Secretary of Defense in 1971. Two objectives of the program are to provide modern tactical communications capabilities to the Services, and to achieve interoperability of DoD communications systems. [4] Major General John E. Hoover, Director of the TRI-TAC Office, recently summarized the focus of the effort as follows:

> By developing a digital system.....we are providing a new capability in tactical communications. We will supply better service to the final user; we will be better able to cope with the growing requirements for transfer of data; we will realize the many technical advantages inherent in digital technology, as compared with the analog world; and we will have made the achievement of widespread security economically feasible. [5]

The TRI-TAC Office is nearing completion of a set of planning documents that will define the overall architecture for the future tactical system. The approach to transition is evolutionary, because economic limitations and other constraints inhibit a rapid changeover to the objective system. We consider the TRI-TAC Architecture in more detail in Chapter 3.

The Integrated Tactical Communications Systems Study (INTACS)

INTACS is a major study effort by the U.S. Army to plan the fielding of a modern, mobile, digital, automatic, communications system. The study takes into account the system architecture defined by

the TRI-TAC Office and the hardware (and software) developments now in
progress. The study is aimed at translating the known requirements in-
to a workable communications network. The INTACS study, if implemented
in U.S. Army programs, will dominate the character of the communications
system of the 1980's and 1990's. [6]

Tandem Voice Links

The system approach to transition being developed in the TRI-TAC
and INTACS efforts involves the tandeming of various voice links. The
PCM links now employed in the U.S. Army switched network will be retained
(in some parts of the system) through the decade of the 1980's. Single
channel net radio terminals which use 16 kBit/sec CVSD will be intro-
duced. Digital telephones which use 16 and/or 32 kBit/sec CVSD will also
be fielded.

A narrowband secure voice terminal will be introduced into the
DoD system in the 1980's. Testing is being completed now "by the Nar-
rowband Secure Voice Consortium to pick a DoD-wide narrowband digital
voice technique." [7] (The use of the term narrowband relates to the digi-
tal channel rate, i.e., a narrowband technique is one with channel rate
less than 5 kBits/sec. Such a digital signal may be conveyed in an
analog telephone channel.). The candidate narrowband terminals are vo-
coders, many of which employ the linear predictive coding (LPC) approach.
To simplify the discussion which follows, let us assume that an LPC vo-
coder will be fielded. A likely bit rate choice is 2.4 kB/s, so we label
the terminal (and the A/D technique) as "LPC-2.4."

The transitional system of the 1980's will involve tandem connec-
tions of PCM, CVSD, LPC-2.4 and analog links. Thus, a question of vital
importance is the performance of digital voice links.

COMPARISON OF DIGITAL VOICE LINKS [8]

The key factors in the choice of an A/D conversion technique for voice are performance, channel rate, and cost. In this section, we contrast PCM, CVSD, and LPC in terms of these factors.

A summary of the comparison is shown below:

	Performance	Channel Rate	Cost
PCM-64	"excellent" (SNR $\approx$ 36dB)	64 kB/s	low
PCM-48	"very good" (SNR $\approx$ 26 dB)	48 kB/s	low
CVSD-32	"very good" (SNR $\approx$ 25 dB)	32 kB/s	low
CVSD-16	"good" (SNR $\approx$ 16 dB)	16 kB/s	low
LPC-2.4	"fair/good" (intelligibility $\approx$ 90%)	2.4 kB/s	high

The adjectives shown represent the subjective judgement of the Author. We examine performance in greater detail in Chapter 4.

We have adopted the notation that "CVSD-16" represents a CVSD A/D converter operating at 16 kB/s, or one digital link of 16 kB/s rate, terminated with CVSD.

PCM delivers "excellent" performance of high SNR, and the cost is relatively low. PCM-64, such as that used in the Bell System, delivers performance essentially equivalent to that of an analog telephone channel. Such a link is quite versatile in analog switched network applications because many links may operate in tandem to compose a circuit of high quality. In addition, non-voice signals (such as modem signals) may be conveyed. Thus, PCM-64 is suitable for application in a general

purpose analog network as a trunking mechanism. PCM-64 is employed in the Bell System and is being fielded in the Defense Communications System (DCS).

The one disadvantage of PCM is the high channel rate, a rate too high to be conveyed on some of the essential channels in the DoD system (e.g., analog telephone and HF channels, and some power-limited satellite channels). PCM does require a relatively error-free channel to deliver satisfactory performance (the bit error rate (BER) should be maintained below 10^{-4} or 10^{-5}).

CVSD provides "good/very good" performance at relatively low cost for channel rates of 16 and 32 kB/s. The quantization noise is noticeable (especially at 16 kB/s), but does not impair effective communication on a one link circuit. CVSD is robust in performance in noisy channels, maintaining high intelligibility for error rates as high as BER = 10^{-1}. The use of CVSD at both 16 and 32 kB/s is planned for the strategic and tactical systems of DoD.

A key uncertainty in the utility of CVSD in a switched network is the ability to tandem with various links. Preliminary testing of tandem links of CVSD at the Electronics Command Laboratories concluded that the maximum number of tandem links (each of which employ CVSD) are as follows:

Channel Rate (kB/s)	Maximum Number of Links
32	5
16	2

These results were obtained under laboratory conditions with no impairments other than the voice A/D-D/A. Other tests suggest that one link of CVSD in tandem with one vocoder link may deliver useful intelligibility.

LPC vocoder performance is labeled as "fair/good" because of the lack of naturalness typical of the vocoder speech. LPC does deliver sufficient intelligibility to be useful on vital command and control links. LPC may be expected to tandem successfully with high quality links such as PCM.

The Narrowband Secure Voice Consortium testing now being documented may provide key answers on the intelligibility of tandem connections of various voice links.

ANALOG AND DIGITAL LINKS

In this section we relate the comparison of digital voice links to the discussion of analog and digital transmission in Chapter 1. Several definitions are given that will be useful in our examination of digital performance in a transitional system.

Signals may be coded in either analog or digital form. Speech is inherently an analog signal. If a speech signal is to be encrypted, it must be converted to digital form, i.e., to a digital voice signal. In a communications network, the voice signal may be conveyed on an analog path, a digital path, or on a tandem connection of two or more such paths. Let us define a link to be an interval along the path through the network over which the message signal remains unchanged in form.

The terms loop and trunk relate to the topology of the network. A loop is the connection between a subscriber and an access switch. A trunk is a connection between two switches. A circuit is the connection between subscribers. Thus, a circuit is composed of two loops, and--if the two subscribers do not share the same access switch--one or more trunks.

A circuit may be composed of one or more links. If a circuit has

two or more links, the path is called a <u>tandem</u> connection. Tandeming occurs at each intermediate point along the path at which the message signal is converted in form. Thus, a <u>tandem point</u> occurs at each intermediate point along the message path at which the information signal is converted from analog to digital, or from digital to analog. The tandem points define the junction of links.

Notice that the number of links in a circuit between subscribers may be counted by adding 1 to the number of tandem points (intermediate A/D and D/A converters) on the path. The analog link path which connects a D/A to a collocated A/D converter is a link, by our definition. Indeed, an increment of impairment may be imposed on the signal by such a link, e.g., noise, loss, etc. But, assuming proper design and operational conditions, the impairment imposed by such a link is negligible, so we will usually ignore the impairment and not count the connection as a link. We refer to such a connection as a <u>junction link</u>.

A <u>channel</u> is a path in the transmission system, defined in terms of the form of transmission (analog or digital). In an <u>analog system</u>, message signals enter the system in analog form, and are conveyed over analog loops and trunks which are connected by analog switches. Trunks and loops in such a system may be provided by analog transmission channels. (If so, the system is a <u>uniform</u> one, homogenous and "pure" analog). Analog trunks and loops in an analog system may also be provided by digital transmission channels.

Consider a modification of the analog system that results if digital transmission is employed to provide the analog trunks between switches. (In this example, the analog trunk employs a digital channel. Each trunk contains an A/D converter to transform the message signal to be conveyed on the digital channel.) This example describes the current Army

Tactical Communications System (ATACS), which employs 48 kB/s PCM in the TD-660 Multiplexer. (In this example the analog link at each switching node does impose impairment. Most Army switches operate in a two-wire (2W) node, so a 4W - 2W Hybrid is connected in the signal path, resulting in a 3 dB attenuation of the information signal at each switching node. We may observe in Figure 5(b) that the performance of a 48 kB/s PCM trunk is a function of input signal level; SNR may degrade by 0.8 dB for each dB change in input level. In addition, noise impairment may occur as the analog junction link traverses a wire or cable subsystem within the signal center.)

The dual case is a _digital system_, in which message signals access in digital form, and are conveyed over digital loops and trunks which are connected by digital switches. Voice signals are converted to digital at the point of system access, and back to analog at the point of egress. A digital system is uniform if subscribers are connected by one digital link. (No tandeming occurs in a uniform system.)

During the two decades of transition toward a predominantly digital DoD system, a _hybrid_ mix of both analog and digital subscribers, loops, trunks, and switches will exist in the network. At the beginning of the transition, the system is an analog one with both analog and digital transmission. As the transition progresses, an increasing fraction of the subscribers, loops, trunks and switches are converted to digital operation.

Digital links are fully defined by specifying the bit rate and the error rate. Analog links are much more diverse in character, and performance is defined in terms of bandwidth (the amplitude versus frequency characteristic), attenuation, noise of various types (thermal, quantization, cross talk, hum, impulsive, intermodulation, etc.), output SNR as

a function of input signal level (ISL), and nonlinear effects (echo, companding, etc.). Thus, the performance of analog links is difficult to define and results from the cumulative effects of distributed phenomena.

Many types of analog loops and trunks will exist in the transitional system of the 1980's. A partial listing follows:

1. Analog radio and wire channels

2. Digital channels which employ A/D and D/A converters:
   a. 64 kB/s PCM
   b. 48 kB/s PCM
   c. 32 kB/s CVSD
   d. 16 kB/s CVSD
   e. 2.4 kB/s LPC (LPC-2.4)

During the initial years of transition (late 1970's and early 1980's) several additional types of digital channels will exist in the system, such as 2.4 kB/s channel vocoders, 9.6 kB/s CVSD, and 50 kB/s PCM.

DIGITAL VOICE TANDEMING - THE KEY PROBLEM

In this section, we consider a conceptual model of voice tandeming, and present the purpose statement that is the focus of this research.

A Model

A conceptual model of a voice network is shown in Figure 6. Let us consider each shaded sector in the circle to represent digital voice coded by the following methods:

Sector	Digital Voice Technique
1	CVSD-16
2	CVSD-32
3	LPC-2.4
4	PCM-48

The unshaded sectors represent analog voice. (In this discussion we ignore impairment due to analog links and digital channel errors.) Let a capital letter represent the end points of a circuit path between subscribers. An A/D or D/A conversion is performed as the path traverses a
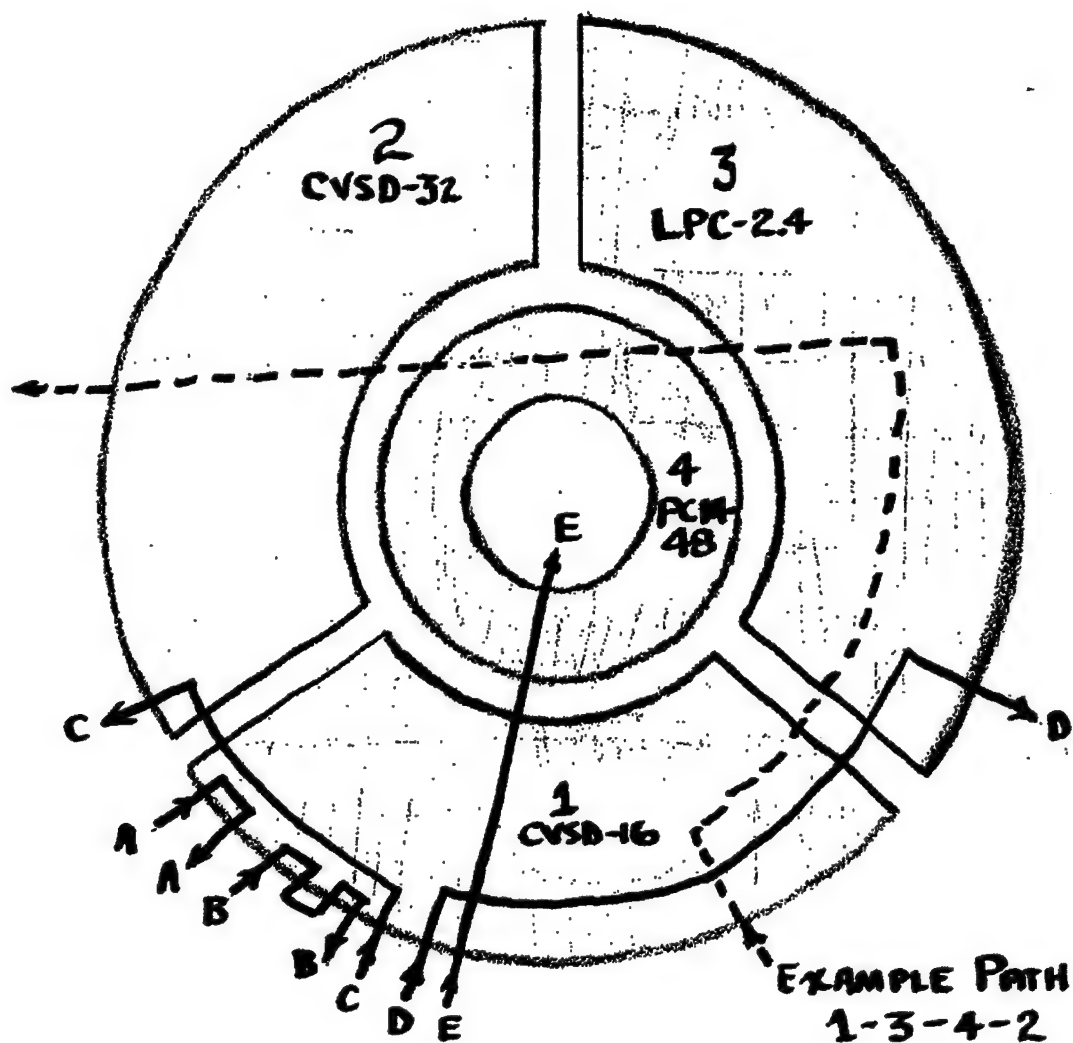
Figure 6. An Abstract Model of Voice Tandeming

boundary line upon entry or exist from a digital sector. The path
through a sector represents one digital link. Passage through an analog
region between sectors is an analog junction link.

Path A illustrates a connection between subscribers of one link
of CVSD-16. Similarly, Path B is composed of two links of CVSD-16. (Note
that each digital link involves one A/D and one D/A conversion at the end
points.)

The model of Figure 6 is useful as an aid to think through the
question posed at the end of Chapter 1: Does the ability to tandem imply
any constraint on network design, or impose any limitations of operational
capability?

Let us consider circuit paths of one, two, three, and four links
that originate from a subscriber of sector 1 (CVSD-16). The only one
link circuit is Path A, and the results summarized above confirm that
Path A is satisfactory. This result confirms the obvious, i.e., if two
subscribers are both equipped with CVSD-16 and if a direct 16 kB/s link
is available in the network, then a satisfactory circuit may be established.
(One may observe that the direct link circuit results in the maximum de-
gree of performance margin, because the connection imposes a minimum of
digital voice conversion distortion. Thus, Path A would remain more
useful in conditions of high channel error rate than paths which include
additional links.)

We summarize an assessment of the two-link connections as
follows:

Satisfactory Paths	Satisfactory Paths of Minimum Margin
C (1-2)	B (1-1)
E (1-4)	D (1-3) (?)

We summarize an assessment of some of the three-link connections
as follows:

Satisfactory Paths	Questionable Paths	Unsatisfactory Paths
1-2-2	1-2-3	1-1-1
1-2-4	1-4-1	1-1-3
1-4-2	1-1-4	1-3-1
	1-2-1	
	1-4-3	

One observation that may be drawn from this assessment follows: A three-
link connection that originates and terminates as CVSD-16, i.e., 1-( )-1,
is either questionable (if 1-2-1 or 1-4-1) or unsatisfactory (if 1-3-1 or
1-1-1). Another view of this observation is that if the network contains
a community of CVSD-16 subscribers who may access the network only through
one link of CVSD-16, then a network connection between any two such sub-
scribers is questionable at best. (This example applies if CVSD-16 net
radio users may only access the switched network at an analog interface.
Two such users are potentially isolated from one another.)

Another observation is that an unsatisfactory three-link path
implies that all four (or more)-link paths that contain the three-link
sequence are also unsatisfactory. For example, that 1-1-1 is unsatisfac-
tory implies that 1-2-1-1, 1-1-2-1, 1-1-1-2, 1-3-1-1, 1-1-3-1, 1-1-1-3,
...(and others)...are also unsatisfactory.

We summarize an assessment of a few of the four-link connections
as follows:

Satisfactory Paths	Questionable Paths	Unsatisfactory Paths
1-2-4-4	1-2-4-1	1-4-1-1
1-4-4-4	1-2-2-2	1-2-1-1

The conclusion to be drawn from these considerations is that the

performance of tandem voice-links may impose limitations of operational capability (i.e., some available network paths are unsatisfactory). Another point of view is that the performance of tandem links does imply necessary constraints on network design.

The assessments offered above are tentative and judgemental. In Chapter 4 we seek to refine the assessments by analysis and by examining recent experimental test results.

Some tentative conclusions are supported by the assessment of the network model.

Tentative Conclusions

The assessments do provide some general conclusions concerning the questions posed at the end of Chapter 1:

(1) What ability do these techniques have to deliver satisfactory performance if operated in tandem link connections?

Links of PCM-64 deliver rugged, versatile performance in tandem circuits, and do convey non-voice signals. Links of PCM-48 are likely to deliver satisfactory performance in most tandem voice circuits, but adding one link of PCM-48 (to establish an N-link path) may render unsatisfactory a marginal connection of N-1 links. CVSD-32 is somewhat versatile in tandem links (but distinctly less versatile than PCM-64), and is of questionable utility for non-voice signals. Links of CVSD-16 are satisfactory if direct, or in restricted tandem circuits with PCM or CVSD-32. But links of CVSD-16 become marginal (or unsatisfactory) in circuits which contain two (or more) links of CVSD-16 and/or LPC-2.4, and circuits which contain CVSD-16 are of little or no use to convey non-voice signals. Links of LPC-2.4 are of limited versatility in tandem circuits, and have essentially no utility for the common non-voice signals. Analog links interleaved in a circuit of digital links may impose additional impairment.

(2) Does the ability to tandem imply any constraint on network design, or impose any limitations of operational capability?

Yes, both constraints and limitations. Network topology, connectivity, and routing control in a transitional, hybrid system must incorporate engineering design features to accommodate the diverse (and not analytically quantifiable) performance of various circuits of tandem links. The sizing of trunk groups is not a simple (common-user) traffic engineering problem, but must take into account the needlines of various user communities-of-interest and the slowly changing mix of terminal equipments and trunking.

Limitations of operational capability are certain to occur in the transitional system because some circuit paths that could be established deliver unsatisfactory performance. (In other words, whether or not a given trunk is useful to a particular subscriber depends upon where he is in the network, who he is calling, the terminal equipments of both subscribers, the connectivity possibilities of the current network, and various random influences such as network loading, and noise and alignment conditions on analog links.) "Common-user" networks that must route traffic based upon segmented communities do not gain the full measure of economy of scale as unsegmented networks.

The Problem.

No single measure of performance of analog and digital voice communications links is known that is sufficient for engineering design of a network composed of diverse links. Experimental testing is necessary to validate the usefulness of a user-to-user connection composed of tandem links.

A two-decade transition of DoD communications systems to digital

operation is beginning. Secure voice connections in the "transitional DoD System" of the 1980's will involve tandeming of various voice links (e.g., analog, pulse code modulation, delta modulation, and vocoder links). The transitional system design will determine the number and nature of tandem links traversed by the network paths between every pair of users in the system. Thus, the system design may result in groups of users that are isolated from certain other users in the network.

The purpose of this research is to determine if the U.S. Army tactical communications system design for the 1980's includes isolated users. The research approach is to examine plans for DoD systems in the 1980's and analyze a model of the "1980's transitional system" using experimental and analytical, tandem voice-link, performance data. If isolated users exist, system design improvements will be explored.

CHAPTER END NOTES

1. Thomas C. Reed, "Command and Control and Communications RDT&E," Signal Magazine, October 1975, p.7.

2. Some protection of voice communications is accomplished by bulk encryption, a technique that provides encryption of multichannel radio transmission signals on selected radio hops. (Bulk encryption capability is being fielded in the ATACS.) Such protection is not equivalent to security, because all radio hops on various circuit paths are not encrypted, and because wireline loops, trunks and internal signal center links are not protected. Notice that encryption of a digital link protects against hostile SIGINT (monitoring and decoding the message), and protects against some hostile EW (such as imitative communications deception.)

3. George A. Godding, "Think Tactical," Signal Magazine, September 1975, pp. 16-18.

4. John E. Hoover, "The Joint Tactical Communications Program," Signal Magazine, July 1975, p. 16.

5. Hoover, p. 17.

6. Charles R. Myer, "C^3 in the Forward Areas," Signal Magazine, August 1975, pp. 89-93.

7. Reed, p. 61.

8. E. D. Harras and J. W. Preusse, Communications Performance

of CVSD at 16/32 Kilobits/Second, Final Report - Phase II, U.S. Army

Electronics Command, Fort Monmouth, New Jersey, January 15, 1974,

pp. 13-??.

CHAPTER 3

DIGITAL COMMUNICATIONS PLANS -
THE 1976 APPROACH

In this Chapter we examine the policies and the planning approach to digital communications in DoD that guide, influence, or limit the secure voice communications of the U.S. Army. We begin with a summary of DoD policy guidelines that focus the digital transition toward the 16 kB/s channel rate. In the second section, we survey TRI-TAC Architecture which forms the broad design for future tactical systems. Next, the specific objective system and transition strategy for U.S. Army communications is examined. In the fourth section, we consider interoperability between various regions of the overall DoD system. The Chapter concludes with a summary.

DOD POLICY

The thrust of DoD policy is that future communications systems will be digital, secure or securable, and will employ end-to-end security over 16 kB/s circuits. In 1973, Dr. E. Rechtin, then Assistant Secretary of Defense for Telecommunications (ASDT), summarized the policy approach as follows:

> ....The decisions of the Secretary of Defense and the Chairman of the Joint Chiefs of Staff on how to configure command and control of the SIOP or nuclear forces have made it clear that communications options must exist for clear and unimpeded command and control between the national command authorities and the executing commanders in the field. Such communications are not possible with fragmented and disconnected separate networks designed to different criteria by different groups. An overall communications system architecture is mandatory if the national command authorities are to get the communications they have stated they need....

....It was as a result of recent experience and a consequence of joint discussion between the security people and communications committees that DoD recently decided to procure no further military voice radios unless they were either secured or securable, and, further, that our goal was widespread, end-to-end security. Link-by-link security, although useful as a supplement in high threat areas, is clearly not a substitute for end-to-end security in the world in which we must live.

....We should do something to standardize bit rates for both long-haul and tactical communications. This led to the decision to use 16 Kilobits per second, with an intermediate use of 32 Kilobits per second, for secure digital voice traffic in the field and over appropriate long-haul circuits. It now appears commercial common carriers may offer a 16 Kilobit per second digital service, which should considerably improve our chances for widespread, end-to-end secure voice. [1]

This policy initiative of 1973 served to focus the attention of system engineering organizations in DoD toward a single, specific, common channel-bit-rate for future systems, 16 kB/s. The choice of 16 kB/s represents a compromise between many conflicting considerations that apply to local subsystems in various parts of the overall DoD communications system. The interaction among the Services and DoD agencies since 1973 has resulted in the merging of system design choices into a nearly coherent system architecture. This merging of system approaches is a theme that is apparent in the plans of DoD organizations. In the following sections we examine the TRI-TAC architecture, the Army INTACS study, and other DoD systems.

THE TRI-TAC ARCHITECTURE [2]

Major General John E. Hoover recently summarized the scope and goals of the Joint Tactical Communications Program as follows:

....The program includes tactical, multichannel, switched communications and related COMSEC, access and interface capabilities. The program was established to:
(1) Provide the Services, on a timely basis, modern tactical communications capabilities;
(2) Achieve interoperability within the Department of Defense and with our allies;

(3) Eliminate duplication among Service Research, Development
and Procurement programs; and

(4) Do all of these things in the most economical manner.

....Essentially, TRI-TAC is a single program which provides
for the design, development, and acquisition of the next genera-
tion of tactical communications capabilities for all the Services--
while achieving interoperability among them, eliminating duplication,
and minimizing costs in the process. [3]

The system engineering description of the TRI-TAC Architecture

is detailed in:

....A single architectural document including Systems Objec-
tives, an Implementation Plan, Subsystem Architectures, and Assess-
ment of Technology, Methodology for Design and Analysis, and some
fourteen annexes covering such system-wide aspects as Communica-
tions Security, Traffic Handling, and Network Timing. [4]

The architecture provides the broad design for a digital, common-user,

switched system which provides clear voice and secure voice services,

and data services such as teletype, intercomputer, record, sensor, and

facsimile.

TRI-TAC Development and Acquisition Programs

A useful insight may be gained by examining the development and

acquisition programs for hardware to be fielded during the initial phase

of the transition. A concise summary of the developments is given in Ap-

pendix I. The Appendix contains Tables 1 through 7, which detail the

Phase I (through 1982) hardware to be deployed in the subsystems of the

tactical network. The scheduled dates for the beginning of production

of these equipments occur in the interval April 1976 through November

1981. [5] (Definitions of words, phrases, abbreviations, and acronyms are

listed in the Glossary of the TRI-TAC Architecture. [6])

The switching subsystem improvements during Phase I are listed in

Table 1. [7] The principal improvement is the AN/TTC-39 Automatic Switch-

board. In Phases II and III the AN/TTC-39 continues to be deployed in the

Land Based System. A 16 kB/s switching network will be introduced in the Naval System during Phase II, and a 2.4 kB/s access capability will be fielded. [8] (Phase II is the interval 1982 - 1990; Phase III is 1990 - 1998.)

Static Subscriber Access Subsystem improvements during Phase I are listed in Table 2. [9] The key improvement is the Digital Subscriber Voice Terminal (DSVT). The DSVT is a secure voice terminal which employs CVSD (operating at 16 or 32 kB/s).

Phase II improvements include a new family of 30 to 90 line Unit Level Switches, expansion of message switching access, and fielding of additional multichannel radio assets. "The Army ATACS (TD-660/1065) Analog/PCM multiplex inventory will play a key role in the transition through the hybrid analog/digital environment of Phase II." Phase III improvements include switching, multiplexing and radio transmission. "Phase III will see the phase-out of analog subscribers and thus the Army ATACS "universal" channel equipment." [10]

Mobile Subscriber Access Subsystem improvements during Phase I are listed in Table 3. [11] Phase II improvements include the proposed fielding of a distinct Mobile Subscriber Access (MSA) subsystem with three subsystem components: Mobile Subscriber Control (MSC), Mobile Subscriber Terminal (MST), and Access Unit (AU). [12]

Trunk Transmission (Surface) Subsystem improvements for Phase I are listed in Table 4. [13] Phase II improvements include troposcatter and HF facilities. Troposcatter, SHF LOS, UHF LOS, and HF capabilities would be improved during Phase III. [14]

Trunk Transmission (Space) Subsystem improvements for Phase I are listed in Table 5. [15] Further improvements would be implemented during Phases II and III.

Digital Voice Bit Rates

One design objective for the tactical switched communication systems is end-to-end security.[16] The DSVT is to provide traffic encryption on an end-to-end basis.[17] The modification of the DSVT is to provide a capability for push-to-talk operation an 16 kB/s.[18] "The initial channel rate for TRI-TAC Land-Based Systems is 32 kB/s; the rate in the Naval System will be 16 kB/s."[19]

The TTC-39 Specification provides the following guidance on Analog/Digital Conversion:

> Digital voice terminals shall digitize analog voice using the continuously Variable Delta Modulation (CVSD) technique at a bit rate of 32 kB/s. The TDMX of the circuit switch sybsystem shall operate at a single channel rate of 32 kB/s. ....It shall be possible for any voice subscriber connected to the AN/TTC-39 circuit switch subsystem to be able to communicate with any other voice subscriber.
>
> ....The circuit switch subsystem must, therefore, provide A/D and D/A devices for use in connections between a subscriber using an analog voice terminal and a subscriber using a TRI-TAC digital terminal (32 kB/s, CVSD).[20]

The mature TRI-TAC "objective system" will employ all digital 16 kB/s trunking in a homogenous common user, circuit switched network. Thus, subscriber to subscriber circuits in the mature network are direct digital links. Tandem conversions between digital voice links are essentially eliminated. Thus, in the mature objective system the four regions illustrated in Figure 6 would merge into one region of CVSD-16.

Transition Strategy

Recall the discussion in Chapter 2 of analog and digital voice links in a hybrid system. The key challenge of transition is to provide continuity of user services during a period of gradual evolution of the system from a predominately analog one to a predominately digital one. Digital subscriber terminals, trunks and switches will gradually be phased

into the network. The high cost of fielding a new system dictates a lengthy transition.

The TRI-TAC architecture is a broad "umbrella" that covers a wide range of possible system configuration transitions. Such an approach is necessary because the Military Departments and DCA have different starting points for transition. For example, the current ATACS employs digital transmission, whereas the Air Force system has analog transmission. Thus, converting or adapting ATACS trunking to digital operation will be relatively simple.

Other considerations contribute to the need for a versatile transition strategy. Uncertainties in programming actions, year-by-year budgets, and production schedules compel an approach that is flexible in time phasing. The mobile nature of tactical forces dictate that one may not rely on a specific geographic configuration of similarly equipped units; the strategy of transition must be as flexible as the range of tactical employments.

The transition strategy is to field a hybrid circuit switch (the TTC-39) at major system nodes, and slowly expand digital trunks, loops, and subscriber terminals. The space division matrix (SDMX) of the TTC-39 provides automatic switching of existing analog trunks and loops. The time division matrix (TDMX) provides automatic switching of digital trunks and loops. An inter-Matrix Unit (IMU) provides A/D and D/A conversions to interconnect the SDMX and TDMX, permiting crossover between digital and analog trunks or loops. The crossover function permits connections between analog and digital subscribers, analog trunking on a path between digital subscribers (if a digital path is not available), and digital trunking between analog subscribers. Several such crossovers may be needed to establish a circuit between subscribers, depending on the availability

of digital trunking.

The crossover operation in a hybrid network is illustrated in Figure 7. Part (a) depicts a circuit of four links (digital/analog/digital/analog) connecting a DSVT-equipped digital subscriber with an analog subscriber. The crossover is illustrated symbolically in Part (b). Part (a) of Figure 8 illustrates the use of two analog trunks (links) in a circuit path between DSVT subscribers.

A channel rate of 32 KB/s was selected for digital trunking in the TRI-TAC architecture to permit flexibility in crossover, i.e., flexibility in interlacing digital and analog links. Digital links of CVSD-32 provide improved voice quality and ability to tandem.

Another crossover situation is a tandem conversion between digital subscribers equipped with different terminals. Part (b) of Figure 8 illustrates crossover to interconnect a DSVT subscriber (one link of CVSD-32) with a narrow-band subscriber (one link of LPC 2.4).

THE INTACS STUDY [21]

The Integrated Tactical Communications Systems (INTACS) Study is a major effort to define the U.S. Army communications posture in the 1980's. The Study began in 1971 and was essentially concluded in December 1975. A Communications Systems Requirements Study (COMSR) was conducted to identify the needlines to be satisfied by the network. [22] "The INTACS Study provides the mechanism for transcribing the COMSR's into a workable communications network." [23] Another point of view is that the Study is a system engineering design effort to select the minimum cost system configuration that satisfies the requirements and certain constraints (investment profile, available TRI-TAC hardware, etc.).
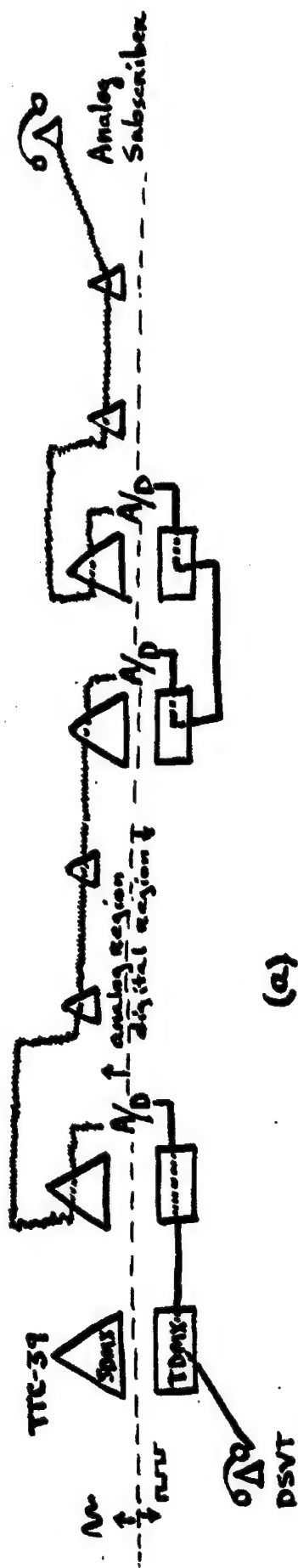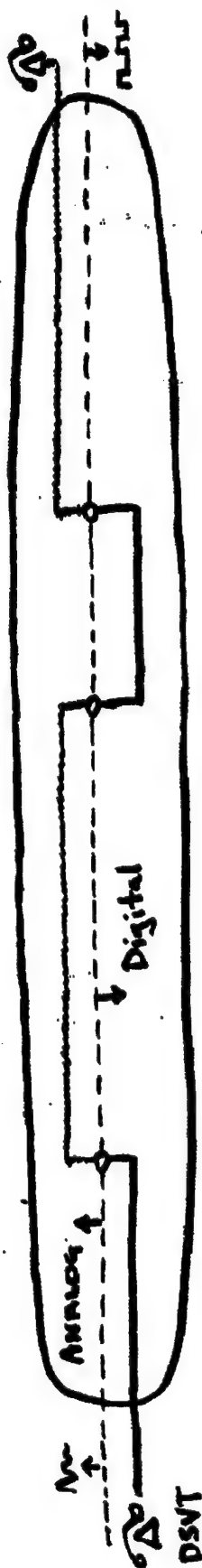
Figure 7. Digital and Analog Links in a Hybrid Network. (a) A four link circuit with two digital links of CVSD and two analog links interleaved. (b) A Symbolic representation of the circuit.
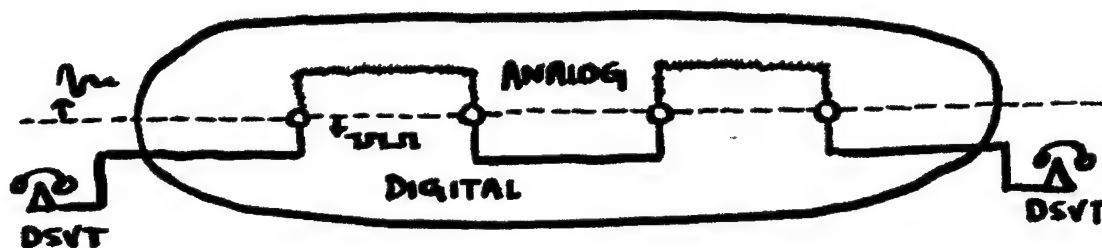
(a) Five Links
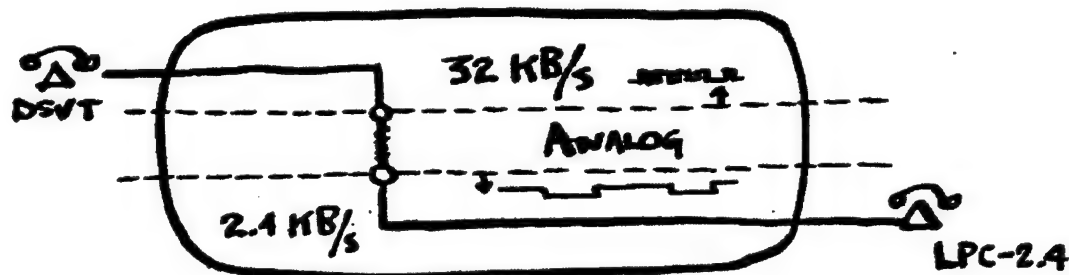
(b) Narrowband Access by LPC-2.4 (CVSD-32
      and LPC-2.4 links joined by an analog
      junction link)

Figure 8.   Tandeming Examples

The INTACS Objective System, the target system for the mid-1990's, is consistent with the TRI-TAC Architecture described above. The multi-channel transmission and switching subsystems are all digital, and digital access is provided for all secure subscribers. Digital trunks in the multichannel system during the transition would operate at 32 kB/s, whereas single channel radio access and MSA subsystems would operate at 16 kB/s. (The analog trunks of PCM-48 would be converted to digital trunks as the transition proceeds.) End-to-end digital operation between all secure subscribers would be possible. (The direct digital link would operate at 16 kB/s in the case of a connection between a 16 kB/s subscriber and a 32 kB/s subscriber. Such operation is accomplished using the Dual-Rate method which is summarized in this chapter.)

The Division Objective System is described in the INTACS Final Report as follows:

The Division Objective System is a highly integrated communications network which is predominantly structured around the TRI-TAC MSA subsystem. MSA integrates the functions of telephones, telephone switching, radio transmission, communication security, radio wire integration, and control into one composite subsystem. Other communications means within the Division area are (1) SINCGARS VHF/FM net radio; (2) LOS multichannel communications and switched wire; (3) Single channel and multichannel tactical satellite communications; and Tactical Record Traffic Terminals, facsimilies, processors, and centers. ²⁵

SINCGARS is the acronym for single channel ground-air radio system. For brevity we adopt the term combat net radio (CNR) to identify single channel radio equipments which employ the CVSD digital voice technique and operate at a 16 kB/s channel rate. SINCGARS is a member of the class of CNR.

The Mobile Subscriber Access (MSA) subsystem proposed for the Division System include six Mobile Subscriber Centrals (MSC) and 275 Mobile Subscriber Terminals (MST). The MST provides automatic, secure

voice user access into the MSA subsystem by radio (using CVSD-16).
The MSC's provide area coverage to serve MST users and CNR access. The
MSC's are interconnected with digital LOS and satellite trunking to other
MSC's and to the multichannel system.

The Corps Objective System is built upon a switched, digital,
multichannel network. Both LOS terrestrial and satellite radio are em-
ployed. (The satellite capability includes Demand Assigned Multiple Ac-
cess (DAMA)). The digital trunking interconnects the nodal TTC-39 circuit
switches. User access is gained by LOS radio and coax cable. The TRI-
TAC Digital Group Multiplex (DGM) equipments are employed. Subscriber
voice terminals include the Digital Subscriber Voice Terminal (DSVT) and
the Digital Non-Secure Voice Terminal (DNVT).

The Theater Objective System is equivalent to the Corps System in
makeup. Increased use of troposcatter is planned at theater level.

Transition Strategy [26]

The transition strategy is outlined in the INTACS Implementation
Plan. Transition is divided into two intervals, the Improved ATACS time
frame (1976 - 1982) and the final period (1983 - 1997). The multichannel
system of the Improved ATACS time frame is essentially that of the current
system. Improvements during the period include increased channel capacity
of LOS systems, digital combining and trunk patching, frequency division
multiple access (FDMA) satellite transmission, automatic small switches,
expansion of some digital trunking capability (use of the TD-1065 with the
TD-660 Multiplex), secure net-radio (the Wideband Security Device - WBSD),
and circuit technical control improvements. [27]

The transition approach during the final period is to field the
TRI-TAC assets beginning at Separate Brigade/Division level, then Corps
level, and then Theater level. The allocation of assets would be guided

by the Department of the Army Master Priority Listing (DAMPL). Management would be applied to the distribution of assets to maintain a near optimum system configuration for high priority users. Improved ATACS assets replaced by TRI-TAC equipments would be deployed to meet the needs of lower priority users. (Thus, ATACS assets would migrate "rearward".)

Summary

The INTACS Study provides a roadmap for transition of Army communications to an all-digital objective system in the 1990's. The Study includes a system engineering design, signal unit force structure, hardware procurement and deployment schedules, and programming and budgeting guidelines.

OTHER SYSTEMS

The objective system configuration and transition approach adopted for several other systems will impact on the ability of subscribers of the Army system to communicate with users in the other systems. Other systems of particular interest are the World Wide Military Command and Control System (WWMCCS), the Defense Communications System (DCS), the U.S. Navy System, and the NATO Integrated Communications System (NICS).

World Wide Military Command and Control System (WWMCCS)

Mr. Thomas C. Reed is the Secretary of the Air Force. In March 1975, while serving as Director, Telecommunications and Command and Control Systems, Office of the Secretary of Defense, Mr. Reed described the WWMCCS as follows:

> The WWMCCS, as now understood, is a composite of command and control capabilities distributed world wide which: (1) Support the NCA by providing the means by which information is received for accurate and timely decisions; apply the resources of the Military Departments; and assign military missions and provide direction to the Unified and

and Specified Commands. (2) Provide for effective command and control support of specific missions of the Unified and Specified Commands and the WWMCCS related management/information systems of other DoD agencies. [28]

The WWMCCS provides secure voice and data services to interconnect the National Command Authorities (NCA) and the commander at the scene of action. A WWMCCS Architecture study effort is in progress which will identify the connectivity requirements (needlines) for the WWMCCS. A WWMCCS System Engineering Office is being organized to accomplish the planning and design necessary to implement the WWMCCS Architecture. [29]

The potential impact of the WWMCCS on Army communications is that WWMCCS needlines will extend into the Army system. If end-to-end security is a requirement on such connections, then the ability to establish direct digital links in the Army system is necessary.

One approach under study for potential implementation for WWMCCS secure voice service is a variable channel rate strategy in which the digital voice terminal would be capable of operating in several modes (at different bit rates). Such operation would permit adapting to a low bit rate when the network is stressed (thus making use of limited transmission capacity) or adapting to a high bit rate to gain performance when capacity is ample. [30] To achieve low bit rate capability, such a terminal would be of cost and complexity comparable to LPC-2.4. [31]

Defense Communications System (DCS) [32]

The DCS is a general purpose, long-haul transmission and switching system which provides a wide range of services to DoD users. The Automatic Voice Network (AUTOVON) is the clear voice "telephone network" of the DCS. The Automatic Secure Voice Communications Network (AUTOSEVOCOM) provides secure voice services to about 1400 subscribers. The Automatic Digital Network (AUTODIN) provides secure record traffic communications.

55

Some insight may be gained into the future DCS secure voice transition from the recent remarks of Brigadier General J. H. Jacobsmeyer, Deputy Director, Plans and Programs, Defense Communications Agency:

> ....By finding it feasible to operate digitally at 16 KBS - at least for secure voice - in the CONUS, we can achieve real convergence of the strategic and tactical communications systems. We should achieve true interoperability and logistics commonality. It is a reasonable objective to foresee any replacement of current overseas DCS switches to be fundamentally the TRI-TAC switches. 35

Lieutenant General Lee M. Paschall, Direct of the Defense Communications Agency recently summarized the approach to interoperability as follows:

> ....it seems certain that the future DCS will be heavily oriented toward interoperability with U.S. tactical communications systems and with the communications systems of our major allies. Interoperability and technical standardization between U.S. tactical communications systems and the DCS will be driven both by the very demanding needs of the flexible nuclear response strategy as well as the requirement to have the capability to precisely control U.S. military forces at the executing level in crisis and contingency operations.

> ....Thus, the strategic and tactical communications systems must have a degree of interoperability and technical standardization such that they are transparent, or very nearly so. 34

The future DCS secure voice system may be summarized as a homogenous 16 kB/s digital switched network which provides direct digital links between CVSD user terminals. Network access in a narrowband mode may be achieved using the LPC-2.4.

U.S. Navy System 35

Some aspects of the secure voice portion of the U.S. Navy System are discussed in the section on TRI-TAC Architecture. Transition of U.S. Navy communications is dominated by the especially challenging transmission channels that are vital in the Naval environment. HF/SSB transmission now in use is a severely bandlimited channel; data rates are constrained to 2.4 kB/s. One the other hand, satellite transmission channels to some key mobile platforms are severely power-limited due to antenna constraints,

so channel rates are restrained to 16 kB/s or less.

The transitional Navy System of the 1980's may be summarized as a 16 kB/s digital switched network which interconnects CVSD subscribers. Direct narrowband links and narrowband access to the digital network will be provided with LPC-2.4. [35]

NATO Integrated Communications System (NICS) [37]

The NICS was initiated in 1970 to seek the integration of various national systems into a common user switching system. "The conceptual NICS is a totally integrated grid network of automatically switched, common user facilities." [38] The ELCROVOX is the current standard NATO vocoder equipment, which operates over the analog links of the switched network. [39]

Summary

The utility of U.S. Army resources to communicate with subscribers of other systems may depend on the system design adopted for those other systems and upon the method of interoperability selected for use at each system boundary. The DCS and the Navy System will employ 16 kB/s digital trunking to provide direct links between CVSD subscribers; both systems will provide narrowband access with LPC-2.4. The ELCROVOX vocoder is employed in the NICS. We examine interoperability between systems in the next section.

INTEROPERABILITY BETWEEN SYSTEMS

In this section, we examine the ability of secure voice users in different regions of the overall DoD system to communicate. We focus on the tandeming problem: "Are conversions to analog required at the system boundaries which join the major regions of the overall DoD system?

A global point of view is adopted in this section. In Chapter 4 we focus on the local considerations of tandeming internal to the U.S. Army tactical system. Tandeming and transition considerations are woven into the same fabric of global and internal architecture.

Regions of the Overall System

Various regions of the secure voice portion of the overall system are shown in the abstraction of Figure 9. The sketch represents a simplified description of the organization of the overall system. Each circular region is a sector of the system within which direct digital links are provided between secure voice users by digital trunking at a given channel rate.

In this discussion we assume that the planning directions summarized in this chapter continue, and that hardware is fielded in the decade of the 1980's in the many regions of the DoD system. The global system abstraction of Figure 9 may be thought of as a system "snapshot" depicting the DoD system of the late 1980's.

We may think of the circular regions of Figure 9 as the "backbone" trunking and switching networks of the DoD system. The region labeled " US ARMY INTACS" represents that sector of the transitional system within which Army DSVT/DNVT subscribers can be connected by a direct link of 32 kB/s trunking. (The method of digital voice A/D conversion associated with these subscribers is shown in parenthesis.) The Army INTACS region is equivalent to the joint region of land-based tactical communications that is equipped with TRI-TAC multichannel transmission and switching assets. Each triangular sector in the sketch represents a user community which may access one or more of the "backbone" systems to establish a long-haul circuit with a subscriber in another community.

The distinction between circular "backbone" regions and the access
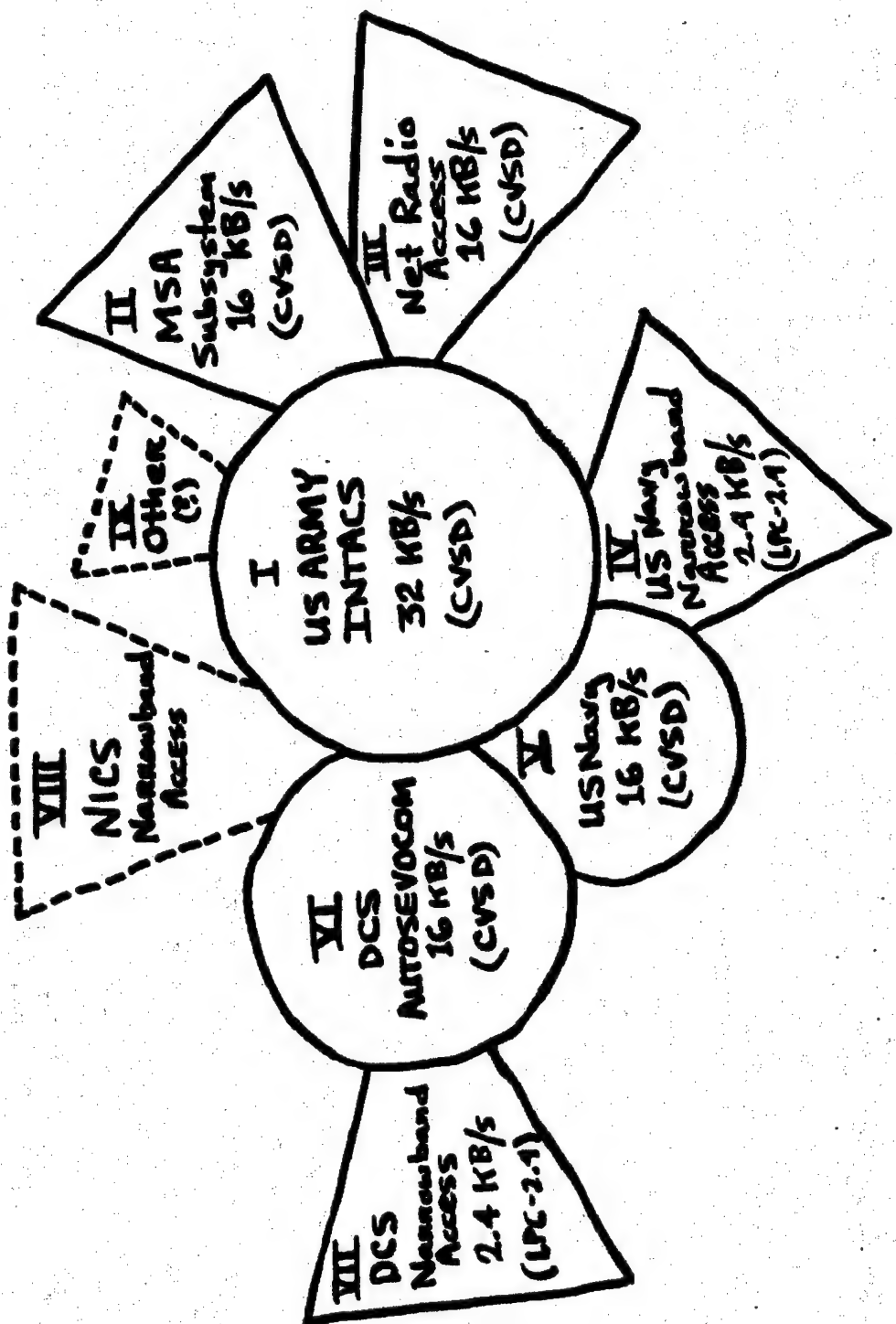
Figure 9. Regions of DoD Secure Voice

regions is that users in a backbone region may establish a direct digital link with any other user of that region. In the access regions, by contrast, some user-pairs may be capable of establishing a direct link, whereas other user-pairs may only be connected by trunking through one (or more) of the backbone regions.

Let us consider the question of tandem links in this "overall system" using the notation employed in the discussion of Figure 6 in Chapter 2. If digital voice signals were converted to analog at the interface boundaries of Figure 9, then a connection between MSA subscribers that is trunked thru the 32 kB/s multichannel network (i.e., a II-I-II connection) would result in a three link circuit (a 1-2-1 circuit in Figure 6.) Such a circuit of tandem links delivers questionable performance. Assuming analog interfaces, an equivalent three link circuit (1-2-1) would result on the following paths:

$$II - I - III$$
$$II - I - V$$
$$II - I - VI$$

$$III - I - V$$
$$III - I - VI$$
$$V - I - VI$$

Thus, a primary factor in determining the degree of interoperability achieved in the overall system is the method selected for interfacing at the region boundaries. (Stated in other words, the choice of interface method determines whether some users are isolated from subscribers in other regions.)

The analog method of interface at the boundaries I-II, I-III, I-V, I-VI, and II-III is listed as one option in the TRI-TAC documents. An interface method called "dual-rate" that avoids such analog interfaces is now being incorporated into various system plans and hardware developments. [40]

The dual-rate method now in planning will permit DSVT equipped subscribers in regions I and VI to establish a connection of one digital link, operating at 16 kB/s. This simple method is accomplished by providing in the DSVT the capability to change to a 16 kB/s rate for operation on those connections that traverse or terminate in a 16 kB/s region. In the switched system, the dual-rate method is performed by automatic "gear-shifting" of the transmission rate at the boundary, i.e., each bit in the 16 kB/s signal is transmitted twice on the 32 kB/s channel. (In the opposite direction of transmission, each pair of identical bits on the 32 kB/s channel is transformed into a single bit in the 16 kB/s channel.)

The dual-rate method is also being implemented for the system boundaries I-II, I-III, II-III, I-V. [42] Thus, if the dual-rate method now in planning is implemented in the hardware fielded in the transitional system, then direct link connections will be possible between all DSVT, KST, and CVR users (i.e., between all secure subscribers with a CVSD digital voice terminal that have system access) in regions I, II, III, V, and VI. A second important implication is that the CVSD equipped subscribers may establish a circuit of two links with narrowband users (i.e., those equipped with LPC-2.4 or MICROVOX) in regions IV, VII and VIII. A circuit between U.S. users would be at worst a (1-3) circuit.

The fact that direct digital links are possible between wideband (CVSD) secure voice users has a key implication for data communications service. Namely, wideband secure voice users may exchange data in a direct mode, from user to user. Thus, the secure voice capability and connectivity of the system may be viewed as a readily available resource for data communications. Such a resource may be useful to satisfy urgent needlines in a flexible, responsive manner.

Summary

In this section we have examined the interoperability of secure voice connections which cross the system boundaries of the overall DoD system. Each region is considered to be homogenous, composed of users equipped with "next-generation" secure voice terminals interconnected with digital trunking.

The dual-rate method is a solution to interoperability at some system boundaries that permits direct digital links to be established between wideband (16 or 32 kB/s) secure voice user pairs. Such operation provides end-to-end digital connections, which makes end-to-end encryption and data transfer possible. The union of these system regions (I, II, III, V and VI) may be viewed as a homogenous 16 kB/s network (i.e., a uniform network.)

We return to the topic of tandem connections which include narrowband links in Chapter 4.

The system level discussion in this section did not include consideration of voice circuits with analog conversions internal to a region (crossover operation). In Chapter 4, we introduce a model of the transitional system to be used to examine circuit performance of various circuits of tandem links internal to the U.S. Army tactical system (i.e., inside the Regions I, II, and III,)

SUMMARY

The thrust of DoD policy is that future telecommunications systems will be digital and provide end-to-end security. A channel rate of 16 kB/s has emerged as the common target of various regions of the DoD system.

The TRI-TAC Architecture is a broad system design which provides flexible system configuration transitions in the tactical networks of the Services. The transition strategy is a hybrid one intended to permit

continuity of user services during the two decades of evolution to a predominantly digital system. The land-based multichannel system employs a 32 kB/s channel rate for digital trunking during the transition. Dual-rate operation permits direct digital links at 16 kB/s.

The INTACS Study describes a specific objective system configuration and transition strategy for U.S. Army communications. The approach is to continue ATACS improvements through 1982, then deploy TRI-TAC developed assets to complete the transition in the late 1990's.

Other systems with which the Army system shares a mutual dependence for design and transition include the WWMCCS, the DCS, the U.S. Navy System, and the NICS. Both the DCS and Navy secure voice systems are based on wideband (16 kB/s) operation, with provision for narrowband (LPC-2.4) access.

Interoperability of secure voice service between systems is based upon the dual-rate method of "gear-shifting" to 16 kB/s those connections which traverse or terminate in a 16 kB/s region of the overall system. Such operation results in transparent, direct digital links between wideband (16/32 kB/s) subscribers.

CHAPTER END NOTES

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    Annex B, "Plan for Implementing the Tactical Switched Communication System Architecture (U)," October, 1975

    Annex C1, "Architecture for Switching Subsystem (Phase I Implementation) (U)", July, 1974

    Annex C3, "Architecture for Mobile Subscriber Access Subsystem (Phase I Implementation (U)," April, 1975

    Annex C4, "Architecture for Trunk Transmission (Surface) Subsystem (Phase I Implementation) (U)," October, 1974

    Annex D, "Assessment of Technology," August, 1974

    Annex F2, "Traffic Handling (U)," November, 1974

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    Annex G, "Glossary," November, 1975

3. John E. Hoover, The Joint Tactical Communications Program," Signal Magazine, July, 1975, p. 16.

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5. TRI-TAC Architecture, Annex B.

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7. TRI-TAC Architecture, Annex B, pp. 5-7, 5-8.

8. TRI-TAC Architecture, Annex B, p. 5-6.

9. TRI-TAC Architecture, Annex B, pp. 6-7 to 6-21.

10. TRI-TAC Architecture, Annex B, pp. 6-6, 6-23 to 6-26, 10-4 to 10-11.

11. TRI-TAC Architecture, Annex B, pp. 7-3 to 7-5.

12. TRI-TAC Architecture, Annex B, p. 7-6.

13. TRI-TAC Architecture, Annex B, pp. 8-4 to 8-7.

14. TRI-TAC Architecture, Annex B, pp. 8-3, 8-8, 8-9.

15. TRI-TAC Architecture, Annex B, pp. 9-12 to 9-18.

16. TRI-TAC Architecture, Annex F2, p. 2-18.

17. TRI-TAC Architecture, Annex B, p. 6-11.

18. TRI-TAC Architecture, Annex B, p. 6-12.

19. TRI-TAC Architecture, Annex C, p. 14.

20. TRI-TAC Office, Specification No. TT-B1-1101-0001, "Performance Specification Central Office, Communications, Automatic, AN/TTC-39 ( ) (V)," Fort Monmouth, N. J., 21 January 1972, p. AVII-8. TDMX is the abbreviation for the time division matrix.

21. The basic reference for this section is the family of INTACS Study documents. A listing of the documents is given in the INTACS

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for the U.S. Army Signal School, Fort Gordon, Ga. The following documents are cited:

    a. Task V Cantidate MidRange Communications Systems

       (1) Volume I, Executive Summary, 16 June 1975.

       (2) Volume II-A, Technical Description, May 1975.

       (3) Volume II-B, Supporting Figures and Tables, May 1975.

       (4) Volume III, Appendices, May 1975.

       (5) Volume IV, Cantidate Evaluations, August 1975.

       (6) Volume V, Objective System, November 1975.

    b. Task VII, Implementation Plan, December 1975.

22. The COMER design includes the ability to update the needline
requirements periodically. Such an update would provide the basis for
modifying the INTACS design.

23. Charles R. Myer, "C^3 in the Forward Areas," Signal Magazine,
August, 1975, pp. 89-93.

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27. INTACS Study, Task VII, pp. 1-2 to 2-6.

28. Thomas C. Reed, "Evolving Strategy - Impact on C^3," Signal
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31. A related observation follows. Such a variable rate terminal implementation may be unable to achieve the size, weight, and power attributes necessary for many portable and mobile tactical applications. In addition, major revision to tactical switching, multiplexing, and transmission designs may be needed to implement a widespread network capable of such variable rate operation. If a region of the tactical system remains as a single-rate circuit-switched network serving CVSD-16 users and the remaining region of the DoD system adopts variable-rate operation, then a tandeming boundary would be created along the interface. On the other hand, tandeming might be avoided except during stressed conditions if one mode of digital voice operation were universally held.

32. Department of Defense, "DCS, A General-Purpose Communications System," Commanders Digest, May 23, 1974, pp. 6-12.

33. John H. Jacobsmeyer, Jr., "Trends in Defense Communications System Switching," Signal Magazine, August 1975, p. 83.

34. Lee M. Paschall, "The Role of the Present and Future DCS in Support of the WWMCCS," Signal Magazine, March 1975, p. 40.

35. Jon L. Boyes, "Telecommunications for a Multiple-Mission Navy," Signal Magazine, January 1975, pp. 6-10.

36. TRI-TAC Architecture, Annex 2, p. 5-3.

37. Harold A. Kissinger, "Organization for Crisis Management in NATO," Signal Magazine, March 1975, pp. 66-70.

38. Kissinger, p. 70.

39. TRI-TAC Architecture, Annex G, p. 25.

40. TRI-TAC Planning Document, "Subsystem Plan for External Inter-
face, "First Draft," March 1974, p. 3-28. (This reference is to be superseded
by Annex G of the TRI-TAC Architecture.)

41. The current description of the dual-rate method was conveyed
by a personal communication with Mr. Michael Norton, Defense Communications
Engineering Center, on February 5, 1976. The dual-rate method is also des-
cribed in an internal DCA memorandum: J. Hammett, Trip Report: "5 to 7
March Meeting with TRI-TAC, Secure Voice Transition Considerations,"
Draft #", 20 March 1975.

42. Personal communication with Mr. Loren Diedrichsen, Chief, Systems
Division, Engineering Directorate, TRI-TAC Office, 4 February 1976.

CHAPTER 4

ANALYSIS - VOICE TANDEMING

In this chapter we focus on the performance of circuits of tandem links internal to the U.S. Army tactical system. (This focus includes Regions I, II, and III in the diagram of Figure 9.) We seek quantitative performance estimates for circuits composed of links of CVSD-16, CVSD-32, LPC-2.4, and PCM-48. The goal is to quantify the performance estimates of circuits for which a tentative assessment was offered in Chapter 2 in the discussion of Figure 6.

A survey of analytical and experimental results is presented, including a simple model for SNR calculations. An analysis of tactical circuits is performed to estimate the SNR performance of likely circuits of up to five tandem links. A brief discussion is given on the extension of tactical circuits into other regions of the DoD system.

TANDEM LINKS - ANALYTICAL AND EXPERIMENTAL RESULTS

In this section we survey the available experimental and analytical results which lend insight into the performance of tandem link voice circuits in the tactical system. The survey includes an examination of U.S. Army PCM, TRI-TAC analysis, ECOM tests and DoD Consortium tests. An arbitrary scale of circuit performance is introduced. Let us begin by examining the PCM-48 currently in use in the ATACS.

U.S. Army PCM

The TD-660 is the PCM Multiplexer employed in the current ATACS.

The use of digital transmission links to interconnect analog switching nodes results in tandem-link circuits. Thus, the tandeming performance of 48 kb/s is of vital interest.

The signal to noise ratio (SNR) performance of PCM-48 was analyzed and measured by Wolfgang H. Fischer at the U.S. Army Electronics Command in 1969.[1] The slot-filter method was used. SNR measurements were taken on PCM-48 circuits established between TD-352 PCM Multiplexers. The TD-352 was the forerunner to the TD-660, and employed a functionally equivalent PCM converter.[2] Thus, we adopt the performance measurements reported by Fischer as the best available data representative of the TD-660 PCM-48.[3]

The measured SNR for PCM-48 is plotted in Figure 10 as a function of input signal level (ISL).[4] We adopt the notation that $\eta(\alpha, \beta)$ is the symbol for signal to noise ratio (the dependent variable). η is a function of ISL--α, and of the channel error rate--β. The measurements did not include channel error conditions, so the results describe only the quantization noise impairment due to variation of ISL. (Thus, we suppress the variable β.)

The maximum SNR (25.5 dB) occurs for an ISL of zero dB ($\eta(0) = 25.5$ dB). Notice that as the ISL decreases, η declines to a minimum of 12 dB for ISL= -20 dB ($\eta(-20) = 12$ dB). (The scale on the abscissa of Figure 10 was adjusted so that the peak SNR occurs at ISL = 0.)

In a subsequent section we compute an estimate of the SNR performance for circuits which include tandem links of PCM-48.

TRI-TAC Analysis

The Error Control Annex of the TRI-TAC Architecture establishes an error budget for TRI-TAC systems and identifies end-to-end error control procedures.[5] The analysis supporting the error control design
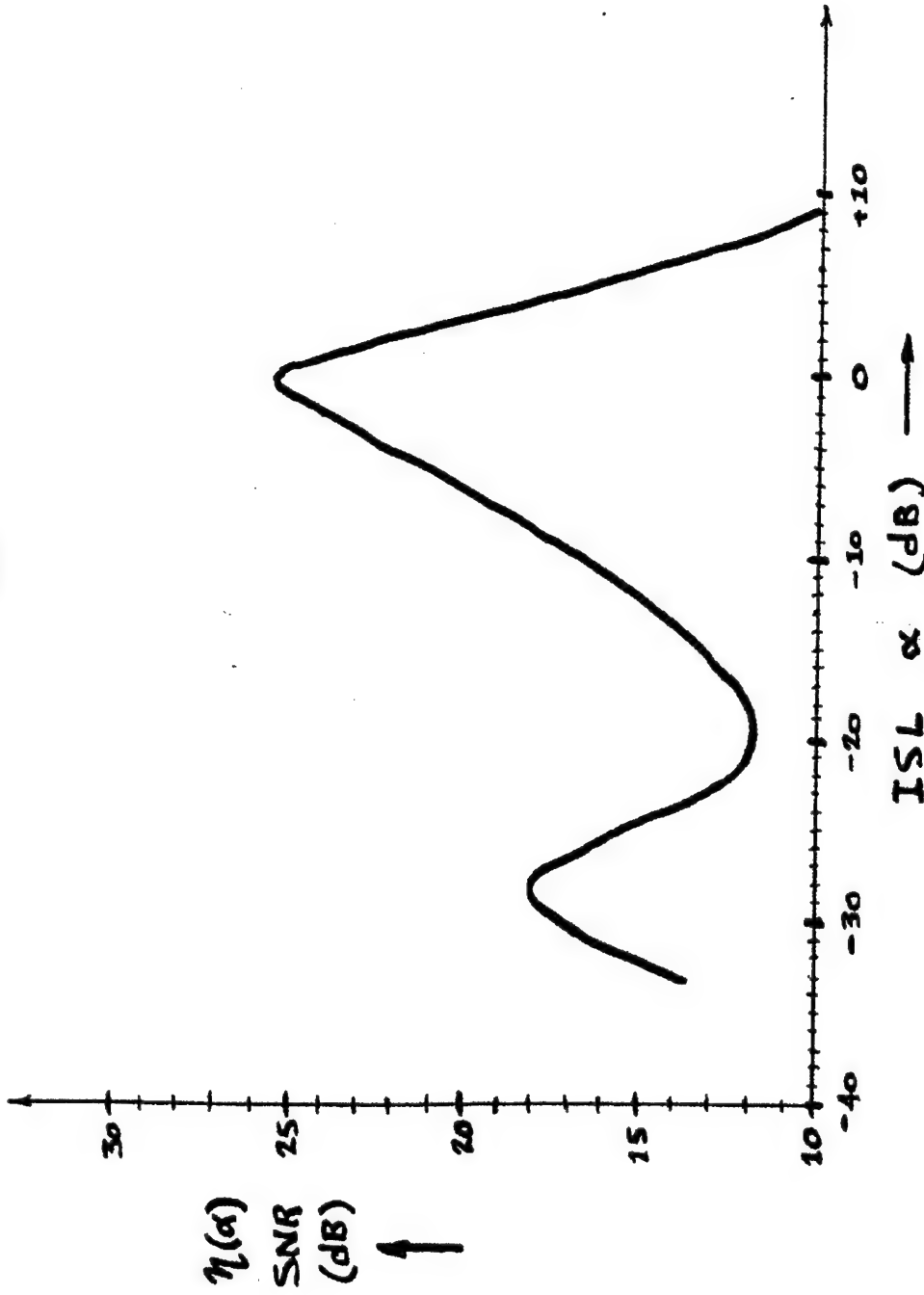
Figure 10   The SNR Performance of U.S. Army PCM-48

choices includes a theoretical examination of the SNR performance of delta modulation on tandem-link circuits with digital channel errors.

An estimate of the SNR performance of one link of linear delta modulation (LDM) is given by the following equation:

$$\eta = 8 + 8\log_2\left(\frac{f_s}{2f_m}\right) \; dB \quad (4\text{-}1)$$

where

η = Signal-to-noise ratio (in dB)

f_s = Sampling rate (in kHz)

f_m = Highest modulating frequency

If we choose f_m = 4 kHz, equation (4-1) becomes:

$$\eta = 8\log_2\left(\frac{f_s}{4}\right) \; dB \quad (4\text{-}2)$$

Evaluating equation (4-2) for sampling rates of 16 and 32 kHz yields the following results:

f_s	η
(kHz)	(dB)
16	16
32	24

Equation (4-2) was derived by Dr. J. B. O'Neal for a rather simple link of LDM. The results may be adopted as an estimate of the actual SNR performance of CVSD. This approach is reasonable because the companding mechanism of CVSD adjusts the step size of the quantizer to deliver CVSD performance that is essentially equivalent to LDM operating with an input signal of optimum level. In other words, the SNR performance of CVSD is near-optimum over a range of ISL of 20 or 30 dB.

Note that equation (4-2) provides an approximation of the SNR performance (due to quantization noise alone) for one link of CVSD operating at a particular sampling (channel) rate. (For delta modulators

the sampling rate -- f_s -- is equivalent to the channel bit rate -- I.) Thus, impairment due to channel errors is not considered.

A modified equation is derived in the TRI-TAC analysis which does account for channel errors. The modified equation provides only approximate results because the noise due to channel errors is computed for a sinusoidal input signal. But the results do provide a useful estimate of the performance of one link of CVSD with channel-error impairment. The SNR performance results are summarized in Table 8. [6]

The results of Table 8 will be used in a subsequent section to compute an estimate of the SNR performance for circuits which include tandem links of CVSD-16 and CVSD-32.

Consider the SNR performance of a circuit composed of several links in tandem. Let η_i be the symbol for the SNR of the i-th link. If we assume that the noise contributions on the several links are uncorrelated, then the noise adds on a power basis and the circuit SNR may be computed as follows:

$$\underline{\eta}_o = \frac{1}{\displaystyle\sum_{i=1}^{M} 1/\underline{\eta}_i} \qquad (4\text{-}3)$$

The bar is added under the symbol η as a reminder that the SNR is given as a power ratio rather than in dB. The following relationships hold:

$$\eta = 10 \log \underline{\eta} \quad \text{dB}$$

$$\underline{\eta} = 10^{0.1\eta} \qquad (4\text{-}4)$$

Equation (4-3) yields the SNR at the output of a circuit of M links, where the SNR of the i-th link is $\underline{\eta}_i$. Equation (4-4) is the relationship between power ratio and dB.

Let us illustrate the use of equations (4-3) and (4-4) with a simple example. Consider a circuit of two links of CVSD-32. Assume

Table 3

SNR Performance of CVSD with Errors

I Channel Rate (KB/c)	Link Error Rate (errors/bit) (%)		SNR (dB)
16	0		16
	10^{-3}	(.1%)	15.9
	10^{-2}	(1%)	15.2
	10^{-1}	(10%)	9.8
32	0		24
	10^{-3}	(.1%)	23.7
	10^{-2}	(1%)	21.9
	10^{-1}	(10%)	12

$\beta = 0$ (i.e., no channel errors). From Table 8, $\eta_1 = \eta_2 = 24$ dB.

$$\eta_1 = \eta_2 = 10^{0.1(24)} = 10^{2.4} = 251.2$$

$$\eta_0 = \frac{1}{\sum_{i=1}^{2} 1/\eta_i} = \frac{1}{\frac{1}{\eta_1} + \frac{1}{\eta_2}}$$

$$= \frac{1}{\frac{1}{251.2} + \frac{1}{251.2}}$$

$$\eta_0 = 125.6$$

$$\eta_0 = 10 \log(125.6) = 21 \text{ dB}$$

Two links of CVSD-32 deliver a circuit SNR of 21 dB, a 3 dB decrease from a one-link circuit. The 3 dB change results from a doubling of the noise power. Similarly, a four-link circuit of CVSD-32 delivers an SNR of 18 dB.

ECOM Tests

A series of tests of the performance of digital voice links were conducted by the Electronics Command in 1973.[7] The purpose of the tests was to verify the utility of CVSD-16 and CVSD-32 for use in the secure voice systems of DoD. The tests involved tandem links of CVSD and vocoders.

The consonant recognition test (CRT) is a test intended to measure the word intelligibility performance of a circuit. The CRT was developed by J. W. Preusse for ECOM and was employed in the 1973 series of tests. Sample test words are played through a circuit and a listening jury records the perceived words. The fraction of correct responses determines the CRT score on a range of zero to 100. The subjective descriptors for CRT scores are listed in Table 9.[8] We note that this scale

Table 9

Subjective Descriptors for CRT Scores

Descriptor	Score
Excellent (E)	Greater than 85
Very Good (VG)	80 to 85
Good (G)	75 to 80
Fair (F)	70 to 75
Marginal (M)	60 to 70
Unacceptable (U)	Less than 60

is an arbitrary one. But the scale is useful to aid in interpreting test results.

Unprocessed speech resulted in a 90.2 CRT. One link of CVSD-32 delivered a 90.5 CRT with no channel errors, degrading to a 90.8 CRT with 10% errors ($\beta = 10^{-2}$). Five tandem links of CVSD-32 with no errors delivered an 81.8 CRT. These results for CVSD-32 fall into the E/VG ranges of Table 9.

One link of CVSD-16 delivered an 84.5 CRT (VG) with no errors, degrading to a 70.3 CRT (F) with 10% errors. Note that these two results compare to SNR estimates of 16 and 9.8 dB respectively. (See Table 9.) Two links of CVSD-16 deliver an 81.3 CRT (VG) with no errors. Adding 1% errors to one of the links degrades performance by about three points, to a 78 CRT (G). A circuit of four links of CVSD-16 delivered a 48.6 CRT (U), a result well below the "unacceptable" threshold. 9

Tests of quasi-analog (modem) signals over a single link of CVSD-16 or CVSD-32 demonstrate that CVSD is not a suitable A/D technique for conveying non-voice (modem) signals. Of five modems tested, only two achieved any degree of success. 10

The tests of CVSD in a tandem connection with narrowband links (such as the channel vocoder) lead to the conclusion that the circuit of two tandem links delivers intelligibility that is essentially equivalent to that of the narrowband system alone. It is clear, however, that the circuit of two links is inferior to a one-link circuit.

DoD Consortium Tests

The Consortium tests have been completed and the report of findings is in coordination draft form at the time of this writing. 11 The testing confirms that a tandem circuit of one link of CVSD-16 and one link

of LPC-2.4 delivers acceptable performance if channel error rates are suitably low. Circuits of three links composed of CVSD-16 and LPC-2.4 deliver marginally unacceptable performance.

The tests represent the most exhaustive investigation of the performance, cost, and complexity of narrowband digital voice processors ever conducted. The test report should receive careful study as a source of insight into the performance of circuits of tandem link.

An Arbitrary Scale of Performance

To simplify the discussion of the next section we adopt an arbitrary scale of performance shown in Table 10. Such a scale is subjective, and represents the opinion of the Author. To emphasize the arbitrary nature of the scale, adjective descriptors were selected that are not commonly used to describe precise quantities.

A "Quality" ("Q") circuit is one with SNR greater than 17 dB. A typical circuit in the Bell System would deliver an SNR of 30 dB or higher. Two links of CVSD-32 deliver a 21 dB SNR, and an 98.2 CRT (Excellent on the CRT Scale of Table 9). A "Q" circuit provides a very comfortable perception of quality and presence to the user, the talker may be readily identified, and high intelligibility is maintained. The circuit has a wide range of margin. That is, considerable additional impairments such as acoustic background noise, channel errors, and level offsets may be suffered, but the circuit remains useful.

A "Suitable" ("S") circuit is one that is generally useful for military voice communications. For example, near the top end of the range is CVSD-16, SNR = 16 dB, CRT - 94.5 (VG). At the low end of the "S" range is a circuit of two links of CVSD-16, SNR = 13 db, CRT - 81.8 (VG). (One may observe that the "Suitable" range on the arbitrary SNR scale corresponds roughly to the "Very Good" range on the CRT Scale.) Circuits in

Table 10

An Arbitrary Scale of Performance

Descriptor	Range of SNR
"Satisfy"	$\eta > 17$ dB
"Marginal"	$13 < \eta \leq 17$ dB
"Tenuous"	$9 < \eta \leq 13$ dB
"Useless"	$\eta \leq 9$ dB

the "S" -range possess some margin to suffer additional impairments.

A "Tenuous" ("T") circuit is usable with some difficulty. Users must adjust their speech patterns to speak slowly and distinctly. Some repetition is necessary. Users are unlikely to identify the other talker. Such a circuit will not be pleasing to most users. The user's ability to perceive the nuances and emotional content of speech is severely impaired. The circuit has little or no margin to remain usable if additional impairment is suffered. Although "Tenuous", such a circuit is usable and represents a worthwhile secure voice resource.

The scale is arbitrary, judgemental, and thresholds should not be considered precise. SNR measurements and analytical estimates display variations between methods. Some pairs of users may communicate successfully on a circuit of SNR somewhat less than 9 dB, while other pairs of users may find a 10 dB SNR circuit to be unusable. Noting these qualifications, we employ the scale of performance to aid in an examination of the relative merit of circuits of tandem links.

ANALYSIS OF TACTICAL CIRCUITS

In this section we present a tandeming model of the tactical system that is useful to aid in visualizing the many possible combinations of circuits of tandem links. An inventory of tactical circuits is accomplished and analysis is performed to estimate SNR performance. Impairments due to ISL offsets and channel errors are examined by considering selected examples.

A Tandeming Model

We seek to examine the performance of typical circuits in the tactical system during the transition of the 1980's. The segmented "tennis-

ball" diagram of Figure 4 provided one abstract perspective of voice tandeming. The diagram of Figure 11 displays a simplified configuration of the tactical network in a theater of operations.

Figure 11 represents a simplified "snapshot" of a system layout that is reasonable for a two-Corps theater in the late 1980's. We assume that steady progress will be made in fielding tactical assets to realize the INTACS Objective System.

The diamond-shaped nodes represent nodes of the MSA subsystem, a 16 kB/s network in the division area serving MST and CNR subscribers. The rectangular nodes represent the digital switching matrix of the Unit Level Switch (ULS) or the TTC-39. The circular nodes represent 4-wire analog switch nodes (e.g., the SDMX of the TTC-39, or the TTC-38). The triangular nodes represent manual analog switches (2 Wire operation). The connections between nodes are dotted or wavy. The dotted connections represent digital trunks between digital switches. (Note that no conversion to analog is required along a dotted path connecting digital switches.) The wavy connections represent analog trunks. (In general, the analog trunks are provided using the PCM-48 A/D and D/A converters of the TD-660 Multiplexer. The transmission is digital, but the trunks are analog ones connecting analog switches.) The slice-of-pie shaped fans represent external paths to NATO or the DCS.

Figure 11 is useful to assist in visualizing the various circuit paths through the network. A catalog of various circuits of tandem links may be obtained by inspection. The Corps on the left, for example, is equipped with TTC-39 switches, unit level switches, the MSA subsystem, digital trunking, and DSVT secure terminals. CNR, MST, and DSVT subscribers may be connected by one digital link if a digital trunking path is available in the network. If a connection between such users is accomplished

LEGEND

◊ MSA Node

▢ Digital Switch

○ Analog Switch (4W)

TTC-39

△ Manual Analog Switch (2W)

----- Digital Trunks
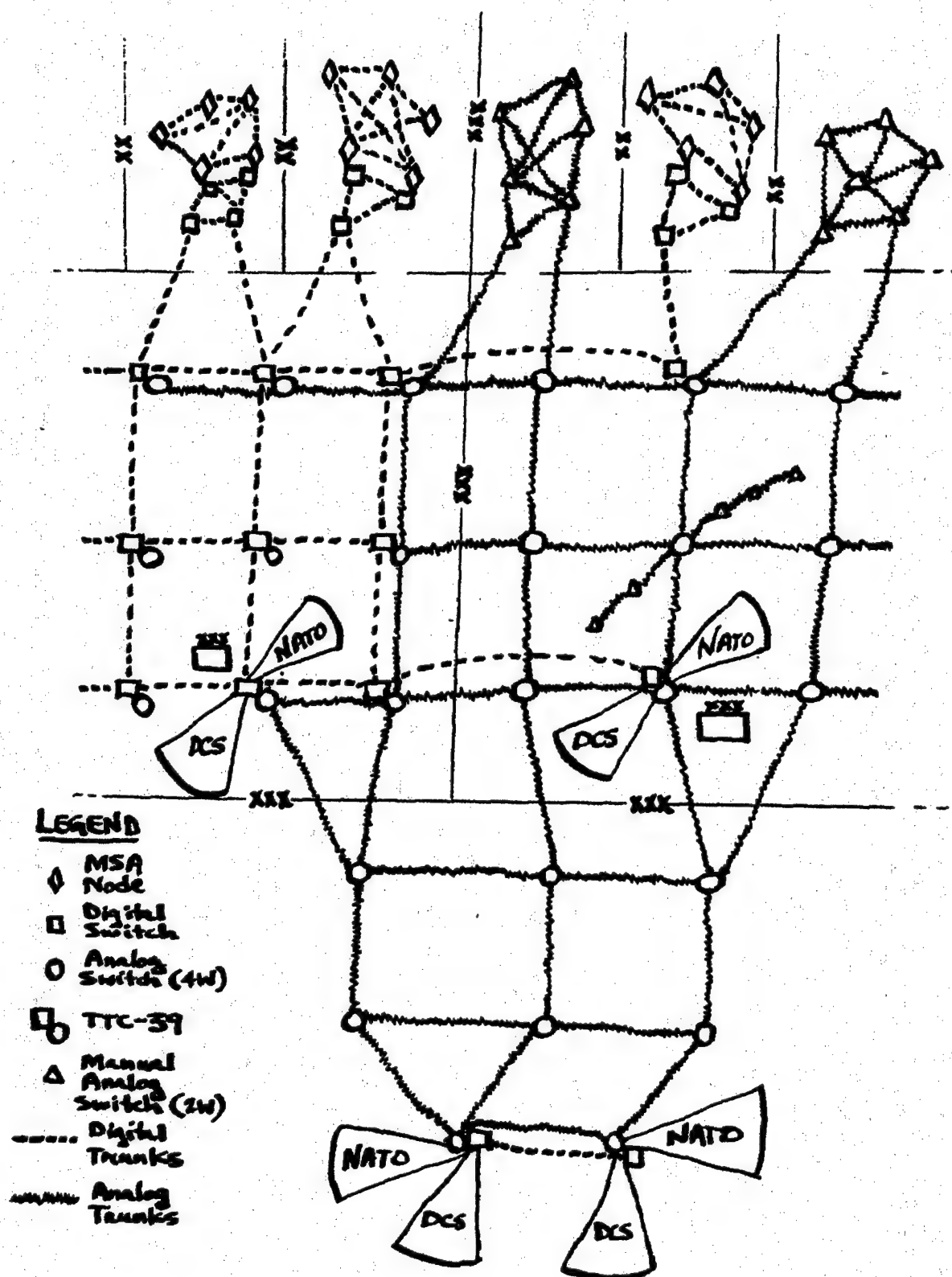
~~~~~ Analog Trunks

Figure 11   A Tactical Network Model

by crossover to analog trunks, a tandem circuit results such as that illustrated in Figure 8a.

The Corps on the right side of Figure 11 is equipped predominately with ATACS assets (analog trunks and switches). Tandem links of PCM-48 occur in this region.

The rectangular and diamond-shaped digital nodes may be interconnected with digital trunking by satellite.

In the next subsection we examine circuit examples.

Tactical Circuit Examples

In this subsection we examine typical circuits internal to the tactical network of Figure 11. We consider circuits with up to six tandem links of PCM-48, CVSD-32, and CVSD-16. The SNR performance of such links detailed in the previous section is applied in this section to estimate circuit SNR performance. Channel errors and ISI offsets are considered in specific circuit examples.

We continue the notation used in the discussion of the "tennis-ball" model of Figure 6 in Chapter 2. A number represents one link as follows:

	Type of Link
1	CVSD-16
2	CVSD-32
3	LPC-2.4
4	PCM-48

In this manner, (1-2) represents a circuit of two links -- one of CVSD-16 and one of CVSD-32. In this subsection we do not consider links of LPC-2.4.

As an introduction to the methods of this subsection, let us consider tandem links of PCM-48. Figure 12 displays the SNR at the output of a circuit of M links of PCM-48, for $M = 1, 2, \ldots, 6$. Three conditions

Figure 12. Tandem-Link Performance of PCM-48

are considered.

The upper curve is a plot of the SNR performance assuming 4-wire switching. (No ISL offset obtains.) The SNR for M links is computed using equation (4-3). From Figure 10, the SNR for each individual link is 25.5 dB.) For this ideal situation, performance remains "Quality".

The middle curve is plotted for 2-wire switching, for which a 3 dB decrement in ISL occurs at each switching node. The SNR operating points for each link are obtained from Figure 10, and the output SNR computed using equation (4-3). Notice that the performance for 5 and 6 links declines into the "Tenuous" region.

The lower curve is plotted in a similar fashion for 2-wire switching and an additional 3 dB loss in ISL at each node. (This condition is a net decrement of 6 dB in ISL at each node. This situation would occur if a wireline loss of 3 dB exists in each switching node.) Notice that performance declines to "Tenuous" for 4 links and "Useless" for 5 or more links.

This discussion of Figure 12 demonstrates the importance of ISL variations in tandem links of PCM-48. The performance of PCM-48 may represent a key limitation of tandem-link performance in the 1980's

Consider the circuit of two links of PCM-48 interconnected by a wireline trunk with 15 dB loss (e.g., a 6 mile wireline). Refer to Figure 10. The first link of PCM-48 has SNR = 25.5 dB. Assuming 2-wire switching, the ISL at the input of the second link of PCM-48 is -18 dB, so SNR = 12 dB. The output SNR is 11.8dB, a "Tenuous" circuit. This example is contrived to illustrate a worst-case situation. Notice that the talker could shift the operating point on the second link to a more favorable position by speaking more loudly, but such action shifts the operating point of the first link down the other side of the performance

peak. If such a shift were accomplished so that the SNR on the two links are equal (at SNR = 15 dB), then the output SNR is 13 dB.

One-Link Circuits. The circuits of one link have already been discussed. Let us summarize.

Users connected by one link of PCM-48 enjoy a "Quality" circuit. (See Figure 10.) The peak SNR is 25.5 dB, and performance declines to 12 dB if the ISL is decreased by 20 dB from the nominal. But on a one-link circuit the talker may simply speak louder or closer to the microphone to shift the SNR operating point into the "Quality" region. PCM is severely degraded by channel errors of .1 to 1%, but multichannel radio links normally provide satisfactory error performance.

Users linked by CVSD-32 also enjoy a "Quality" circuit. SNR = 24 dB with no errors, and performance declines only to SNR = 21.9 dB for error rates as high as 1%. The link remains a "Quality" one over a wide range of ISL due to the adaptive companding of CVSD. Performance degrades to "Tenuous," SNR = 12 dB for a 10% error rate.

Users linked by CVSD-16 obtain a "Suitable" circuit of SNR = 16 dB. Errors degrade performance to SNR = 15.7 dB for a 1% error rate, and to SNR = 9.9 dB for a 10% error rate. Performance is essentially unchanged over a wide range of ISL.

Two-Link Circuits. We consider combinations of three types of links (CVSD-16, CVSD-32, and PCM-48) arranged in tandems of two links. There are

$$\frac{(3 + 2 - 1)!}{(3 - 1)! \, 2!} = 6$$

such combinations. The tandem combinations are listed in Table 11.

The first column of Table 11 is a sequence number for ease of cross reference. The second column lists the composition of links in

Table 11

Circuits of Two Links

#	Circuit Composition	Is Such a Circuit Likely?	Output SNR dB (Descriptor)	CRT if Known
1	(1-1)	No ?	13 ("T")	81.8
2	(1-2)	No ?	15.4 ("S")	84.1
3	(1-4)	Yes	15.5 ("S")	-
4	(2-2)	No	21 ("Q")	88.2
5	(2-4)	Yes	21.7 ("Q")	-
6	(4-4)	Yes	22.5 ("Q")	-

each circuit. The third column contains a judgement as to whether such a circuit is likely in the tactical system. (A question mark entry denotes uncertainty.) The fourth column contains the output SNR performance for the circuit (and the shorthand description in parenthesis). The fifth column lists the CRT score, if available from the ECOM tests. Note that the performance listed in Table 11 is nominal; impairment due to channel errors and TSL offsets are not considered.

Circuit #1 yields performance at the ("T"/"S") boundary (SNR = 13 dB). Such a circuit may be avoided in the tactical system if a digital interface is accomplished between CNR and KSA subsystems, and if the dual-rate method is implemented at the 32 kB/s multichannel system boundary. (As was noted in Chapter 3, if these two conditions are met, then direct digital links are possible between all DSVT, NST, and CNR users -- provided that a digital trunking path is available. The proviso may be met by allocation of digital transmission resources.) We conclude that circuit #1 should be avoided, if possible (by directly connecting users with one link). (Circuits #1 and #4 may occur in conference connections if an analog bridge is employed.)

Circuit #2 yields SNR = 15.4 dB ("S"). Such a circuit may be avoided if the conditions discussed for circuit #1 are met. A one-link 16 kB/s connection would obtain.

Circuit #3 (SNR = 15.5 dB) will occur on connections between CNR/ NST users and analog users that are one PCM-48 trunk distant from the analog interface point.

Circuit #4 (SNR = 21 dB) should not occur in the tactical system, because an intermediate conversion to analog is unnecessary (and undesirable) on a digital path. The connection should be accomplished as a direct link. (Conferencing is a possible exception.)

Circuit #5 (SNR = 21.7 dB) will occur on connections between DSVT users and analog users (that are one PCM-48 trunk distant from the analog interface point).

Circuit #6 (SNR = 22.5 dB) is a common one between analog users.

To summarize Table 11, in the absence of additional impairments all likely two-link circuits are "Suitable" or "Quality." Circuit performance is enhanced if (1-1), (1-2), and (2-2) connections are avoided (i.e., replaced by direct links).

Let us consider two-link circuits that include channel errors and ISL offsets. We select several sets of operating conditions for illustration. The conditions and the results are listed in Table 12.

The sequence number in the first column is keyed to Table 11, and the letter which follows identifies the particular circuit impairment conditions selected for illustration. The channel error rates are listed pairwise in the third column in the same order as the links are listed in the second column. Similarly, ISL offsets are listed pairwise in the fourth column. Output SNR is shown in the fifth column.

Circuit #1a shows that a 1% error rate on each of the links of Circuit #1 leads to an SNR = 12.2, a slight decrease. Circuit #2a has 10% error rate on each link resulting in SNR = 6.8 ("U").

Three-Link Circuits. There are 10 circuit combinations of three-links composed of CVSD-16, CVSD-32, and PCM-48. The performance of these circuits is listed in Table 13.

Circuit #12 will occur on connections between CNR/MST users in different MSA subsystems which are not linked with digital trunking. Such a curcuit has order (1-4-1). [12]

Circuit #15 will occur on connections between DSVT users accomplished on an analog trunk of PCM-48. Such a circuit has order (2-4-2).

Table 12

Two-Link Circuits with Impairments

#	Circuit Composition	Channel Error Rate (%)	ISL Offset (-dB)	Output SNR dB (Descriptor)	
1a	(1-1)	1 , 1	-	12.2	("T")
1b	(1-1)	10, 10	-	6.8	("U")
2a	(1-2)	1 , 1	-	14.4	("S")
3a	(1-4)	1 , 0	0,6	14.1	("S")
3b	(1-4)	10, 0	0,6	9.4	("T")
4a	(2-2)	1 , 1	-	18.9	("Q")
4b	(2-2)	10, 1	-	11.6	("T")
5a	(2-4)	1 , 0	0,6	18.1	("Q")
6a	(4-4)	0 , 0	0,6	19.3	("Q")
6b	(4-4)	0 , 0	0,12	11.8	("T")

Table 13

Circuits of Three Links

#	Circuit Composition	A Likely Circuit?	Output SNR dB (Descriptor)		CRT if Known
10	(1-1-1)	No	11.2	("T")	-
11	(1-1-2)	No	12.7	("T")	79.1
12	(1-1-4)	Yes	12.8	("T")	-
13	(2-2-1)	No ?	14.8	("S")	81.8
14	(2-2-2)	No	19.2	("Q")	87.1
15	(2-2-4)	Yes	19.7	("Q")	-
16	(4-4-1)	Yes	15.1	("S")	-
17	(4-4-2)	Yes	20.1	("Q")	-
18	(4-4-4)	Yes	20.7	("Q")	-
19	(1-2-4)	Yes ?	15.0	("S")	-

To summarize the results given in Table 13, we conclude that of the six likely three-link circuits, three are "Quality" (and thus have wide margin), two are "Suitable" (some margin), and one is "Tenuous" (little margin).

Selected examples of three-link circuits with impairments are given in Table 14. The impairments degrade output SNR.

Four-Link Circuits. There are 15 circuit combinations of four links. The performance of these circuits is listed in Table 15. We conclude that of the six likely four-link circuits, three are "Quality", two are "Suitable", and one is "Tenuous". Circuit #41 would occur in the order (1-4-4-1). The circuit is an extension of #12, the likely three-link circuit with "Tenuous" rating. We note that circuits #12 and #41 would not occur if digital trunking is provided between CNR/MSA user communities.

Five-Link Circuits. There are 21 circuit combinations of five links. The performance of selected five-link circuits is listed in Table 16. We conclude that of the eight likely five-link circuits, four are "Quality", two are "Suitable", and two are "Tenuous". The two "Tenuous" circuits (#55 and 56) are extensions of the "Tenuous" four-link ones discussed above. The same conclusion follows:

Summary. In this subsection analysis was performed to establish SNR performance of circuits of tandem links in the tactical network. Almost all likely circuits of tandem links provide "Suitable" or "Quality" performance, so margin is available to accommodate additional impairments not included in the analysis which may occur in an operating system. The circuits of "Tenuous" performance may be avoided by providing digital trunking between MSA communities. The "twin-peak" SNR curve of PCM-40

Table 14

Three-Link Circuits with Impairments

#	Circuit Composition	Channel Error Rate (%)	ISL Offset (-dB)	Output SNR dB (Descriptor)
12a	(1-4-1)	1 , 0 , 1	–	12.0 ("T")
12b	(1-4-1)	10, 0 ,10	–	6.7 ("U")
12c	(1-4-1)	1 , 0 , 1	0, 6, 0	11.6 ("T")
14a	(0-4-1)	0 , 0 ,10	0, 6, 0	9.3 ("T")

Table 15

Circuits of Four Links

#	Circuit Composition	A likely Circuit?	Output SNR dB (Descriptor)	SNR if Known
30	(1-1-1-1)	No	10.0 ("T")	48.6
31	(1-1-1-2)	No	10.9 ("T")	-
32	(1-1-1-4)	No	11.0 ("T")	-
33	(2-2-2-1)	No	14.3 ("S")	72.0
34	(2-2-2-2)	No	18.0 ("U")	-
35	(2-2-2-4)	No	18.2 ("U")	-
36	(4-4-4-1)	Yes	14.7 ("S")	-
37	(4-4-4-2)	Yes	19.0 ("U")	-
38	(4-4-4-4)	Yes	19.5 ("U")	-
39	(1-1-2-2)	No	11.8 ("T")	65.8
40	(1-1-2-4)	No ?	12.5 ("T")	-
41	(1-1-4-4)	Yes	12.5 ("T")	-
42	(2-2-4-1)	No	14.4 ("S")	-
43	(2-2-4-4)	Yes	18.7 ("U")	-
44	(4-4-1-2)	Yes	14.6 ("S")	-

Table 16

Selected Circuits of Five Links

#	Circuit Composition	A Likely Circuit?	Output SNR dB (Descriptor)
50	(2-2-2-2-2)	No	17.0 ("S"/"Q")
51	(2-2-2-4-4)	Yes	17.5 ("Q")
52	(4-4-4-4-1)	Yes	14.3 ("S")
53	(4-4-4-4-2)	Yes	18.2 ("Q")
54	(4-4-4-4-4)	Yes	18.5 ("Q")
55	(1-1-2-4-4)	Yes ?	12.3 ("T")
56	(1-1-4-4-4)	Yes	12.3 ("T")
57	(2-2-4-1-4)	Yes	14.1 ("S")
58	(2-2-4-4-4)	Yes	17.9 ("Q")

is a factor in tandem-link performance if level offsets occur. Conclusions and recommendations are given in Chapter 5.

The analysis in this subsection does not prove that satisfactory performance will be obtained on the circuits examined. Rather, the analysis provides a useful estimate of relative performance.

One important factor not considered in this section is conferencing. If an analog bridge is implemented to accomplish conferencing, then one additional tandem link is included in each conference circuit path. Such operation would result in additional "Tenuous" and "Useless" conference paths. One interesting alternative to analog-bridge conferencing is push-to-talk (half-duplex) operation with a "one-to-many" (4-wire) digital-switching matrix as the conference bridge.

EXTENSIONS OF TACTICAL CIRCUITS

Let us consider the extension of tactical circuits into other regions of the overall DoD System. Recall the discussion of interoperability between systems in Chapter 3.

The preceding section focused on the (U.S. Army) land-based, tactical system (i.e., Regions I, II, and III of Figure 9). Circuits which exit Region I encoded as CVSD-16 may be extended into Regions V and VI without suffering additional tandeming impairment.

Assuming the availability of digital trunking, a CVSD user in Region I, II, III, or IV may establish a circuit of two links with an LPC-2.4 user in Region IV or VII. Thus, system-wide interoperability of secure voice users obtains in Regions I through VII if the (1-3) circuit is satisfactory. Test results indicate that the (1-3) circuit (CVSD-16 in tandem with LPC-2.4) is acceptable if channel errors are not excessive. One may speculate that such CVSD-16 links may be extended

into Regions VIII and IX to establish usable circuits.

We conclude that a viable architecture for DoD secure voice may be based on a 16 kb/s "backbone" system serving CVSD subscribers (the users of Regions I, II, III, V, and VI). Narrowband access by LPC-2.4 users in Regions IV and VII permits (1-3) circuits to be established with CVSD users. If digital "through-trunking" capability were provided, the LPC-2.4 users could be connected by a direct digital link. ("Through-trunking" is an approach similar to "dual-rate", one that retains the 2.4 kb/s LPC signal in digital form for trunking on 16 kb/s trunks.)

CHAPTER END NOTES

1. Hal G. [...] L. Wexler, <u>Distortion and Quantizing Noise Measurements of PCM Multiplexer TD-352 Using the Slot Filter Method</u>, Research and Development Technical Report ECOM-3029, U.S. Army Electronics Command, Fort Monmouth, N. J., March 1969.

2. In a personal communication in 1975, Dr. Fischer confirmed that the PCM/A digital voice converters of the TD-352 and TD-660 are functionally equivalent. He reported that the TD-660 incorporated improvements to refine the alignment of compandor characteristics between A/D and D/A converters. The single three-segment compandor characteristic is employed in both units. One may conclude that a properly aligned TD-352 link yields performance essentially equivalent to that of a TD-660 link.

3. Dr. Johann Holzer of the U.S. Army Electronics Command initiated and supervised the measurements performed by Dr. Fischer. In a personal communication on 21 January 1976, Dr. Holzer reported that there is a First Article Report on the TD-660 (published by the Raytheon Co., dated 17 May 1968). A maximum SNR = 32 dB was reported for ISL = -4 dBm, and SNR = 12 dB for ISL = -27 dBm. The report is not available. The SNR = 32 dB performance appears to be uncommonly high when contrasted to the maximum SNR = 25.5 dB reported by Fischer. Linear PCM at 40 kb/s delivers SNR = 29 dB for noiselike signals. See Seymour Stein and J. Jay Jones, <u>Modern Communications Principles</u> (New York: McGraw-Hill Book Co., Inc.,

1967) p. 209. Compended PCM delivers peak SNR performance inferior to that of linear PCM. Another source reports a maximum SNR = 27.5 dB for compended PCM-48. The measurements reported by Mr. Fischer are the best available data representative of the TD-660 Multiplexer.

4. Fischer, p. 31.

5. Joint Tactical Communications Office (TRI-TAC Office), "Annex F7 - Error Control," Architecture for Tactical Switched Communications Systems, Report No. TT0-ENG-031-75 AN F7, Fort Monmouth, N.J., November 1975.

6. TRI-TAC Office, p. 22.

7. E. D. Harras and J. W. Pruesse, <u>Communications Performance of CVSD at 16/32 kilobits/second</u>, Final Report - Phase II, U.S. Army Electronics Command, Fort Monmouth, N.J., January 15, 1974.

8. Harras, p. A-1 to A-5.

9. Harras, p. 5.

10. Harras, p. 27-32.

11. DoD Narrowband Digital Voice Consortium, <u>Final Report</u>, Coordination Draft, March 1976.

12. The computation of SNR performance of circuits in this chapter is accomplished without regard for the order of the tandem links. A (1-1-4) circuit is considered equivalent to a (1-4-1) circuit. This is true of the computations because the noise is considered uncorrelated link-to-link in the model. In practice, the ordering of

links does influence performance, due to various nonlinear effects.
Such sensitivity to the ordering of links is especially noticeable
in circuits which include one or more vocoder links.

Chapter 5

FINDINGS

In this chapter we detail the conclusions of the research, offer recommendations, and summarize the study.

CONCLUSIONS

This section has five subsections. First, we examine the U.S. Army System. The approach is to interpret the SNR performance results of Chapter 4 to determine if isolated subscribers will exist in the system, and to draw conclusions which relate the occurrence of tandem links to the reliability of voice performance and the capability for data communications.

In the second subsection we broaden our perspective to consider the overall DoD System. The discussion makes use of the SNR performance results of Chapter 4 to examine the question of interoperability between systems introduced in Chapter 3.

In the third subsection we examine secure voice conferencing. The fourth and fifth subsections contain technical and tactical observations respectively.

In the sixth subsection we summarize the conclusions of the study. A discussion of conclusions is given; a concise summary is stated; and the detailed conclusions are listed.

The U. S. Army System

In this subsection we interpret the performance results obtained in Chapter 4 for circuits internal to the U.S. Army system.

Circuits internal to the U.S. Army system of the 1980's will be composed of tandem links of CVSD-16, CVSD-32, and PCM-48. Let us examine the calculated SNR performance results detailed in Chapter 4. The SNR for circuits of two, three, four, and five links are given in Tables 11, 13, 15, and 16 respectively. The numbers of circuits that fall into each of the four ranges on the arbitrary scale of performance are listed in Table 17. The column labeled "Two Links" summarizes the results of Table 11; there are three likely circuits of two links; "Quality" performance is delivered by (2-4) and (4-4) circuits; a (1-4) circuit delivers "Suitable" performance.

Note that the performance results summarized in Tables 11, 13, 15, 16, and 17 provide an estimate of performance due to quantizing impairments alone. That is, impairments due to channel errors, level offsets, acoustic background noise, and noise corruption at analog tandem points are all ignored. Thus, these results may be considered to represent an upper bound on circuit performance.

Various examples of circuits with channel errors and level offsets are given in the discussion of Figure 10, Table 8, Figure 12, Tables 12 and 14, and in the discussion at the bottom of page 94. We may conclude that channel errors, level offsets, and noise contributions at analog junctions are important factors in circuit performance. To aid in the discussion, let us assume that impairment due to these (more or less) unpredictable effects will typically degrade the circuit SNR by 0.5 to 3dB per analog junction link included in the circuit. [1]

Table 17

Distribution of Performance of Likely Tactical
Circuits (of One to Five Links)

Performance Description	One Link	Two Links	Three Links	Four Links	Five Links
"Quality"	2	2	3	3	4
"Suitable"	1	1	2	2	2
"Tenuous"	0	0	1	1	2
"Useless"	0	0	0	0	0

We note that the circuits rated as "Tenuous" in Table 17 are rather likely to become "Useless" due to additional impairments. The "Tenuous" three-link circuit is a (1-4-1) path (#12 in Table 13) with 12.8 dB SNR, so additional impairment of 3.8 dB or more renders the circuit "Useless." The "Tenuous" four-link circuit is a (1-4-4-1) path (#41 in Table 15) with 12.5 dB SNR, so additional impairment of 3.5 dB or more renders the circuit "Useless." The "Tenuous" five-link circuits are (1-4-2-4-1) and (1-4-4-4-1) paths (#55 and 56 in Table 16) of identical 12.3 dB SNR, so additional impairments of 3.3 dB or more renders either circuit "Useless."

Observe that the four "Tenuous" circuits are all variations of the same catagory; that is, two links of CVSD-16 interconnected by combinations of CVSD-32 and/or PCM-48 (i.e., (1-( )-1) circuits). Referring to Table 11 we note that the "Tenuous" two-link circuit (#1) is a (1-1) path (which was judged to be unlikely) with 13 dB SNR, so additional impairment of 4 dB or more would render the circuit "Useless."

The four likely "Tenuous" circuits result in "isolated" subscribers in the system. We define a pair of subscribers as "isolated" if they may only be connected by a circuit which is "Tenuous" in the absence of additional impairments. Such a definition is reasonable because the probability that a "Tenuous" circuit will be rendered "Useless" (due to additional impairments) is unacceptably high. Another point of view is that the performance margin available on a "Tenuous" circuit is not sufficient to provide reliable service to the user. Still another viewpoint is that even if such a "Tenuous" circuit is useable, it should be avoided as a matter of routine. [2] Let us agree to apply the label "isolated" to any pair of users connected by a circuit that is "Tenuous"

due to quantizing noise alone.

A helpful interpretation of "isolated" user pairs may be gained by thinking of the additional impairments as a random process. The additional impairment on a particular circuit is a random variable which is a function of all the independent random phenomena along the signal path that affect performance. [3]

The notion of the random nature of circuit performance is illustrated in Figure 13. The curves represent the probability density function (p.d.f.) $p(\eta)$ for selected circuits of one, two, three, four, and five links (parts (a) through (e) respectively). Each p.d.f. curve illustrates the frequency of occurance of a circuit as a function of SNR-η. The ordinate represents probability of occurance. [4] We may think of $p(\eta)$ as the density function of a given circuit at different times (e.g., a fading circuit, or a circuit that degrades at night); or, we may think of $p(\eta)$ as the density function describing the various possible circuit paths at a particular time.

The curves sketched in Figure 13 illustrate the concept of the variability of circuit performance due to random impairments. (The shape and extent of the curves represent the opinion of the Author. These random impairments are very difficult to quantify, and extremely difficult to measure in a meaningful way. [5]) The circled point at the right-hand extreme of each of the curves is the upper-bound SNR performance which results from quantization impairment alone.

The one-link circuit (of CVSD-16) illustrated in Part (a) of Figure 13 displays a narrow range of variability (i.e., the p.d.f. curve is narrow). This is true because there are no intermediate analog tandem links in the circuit which may contribute analog impairments.
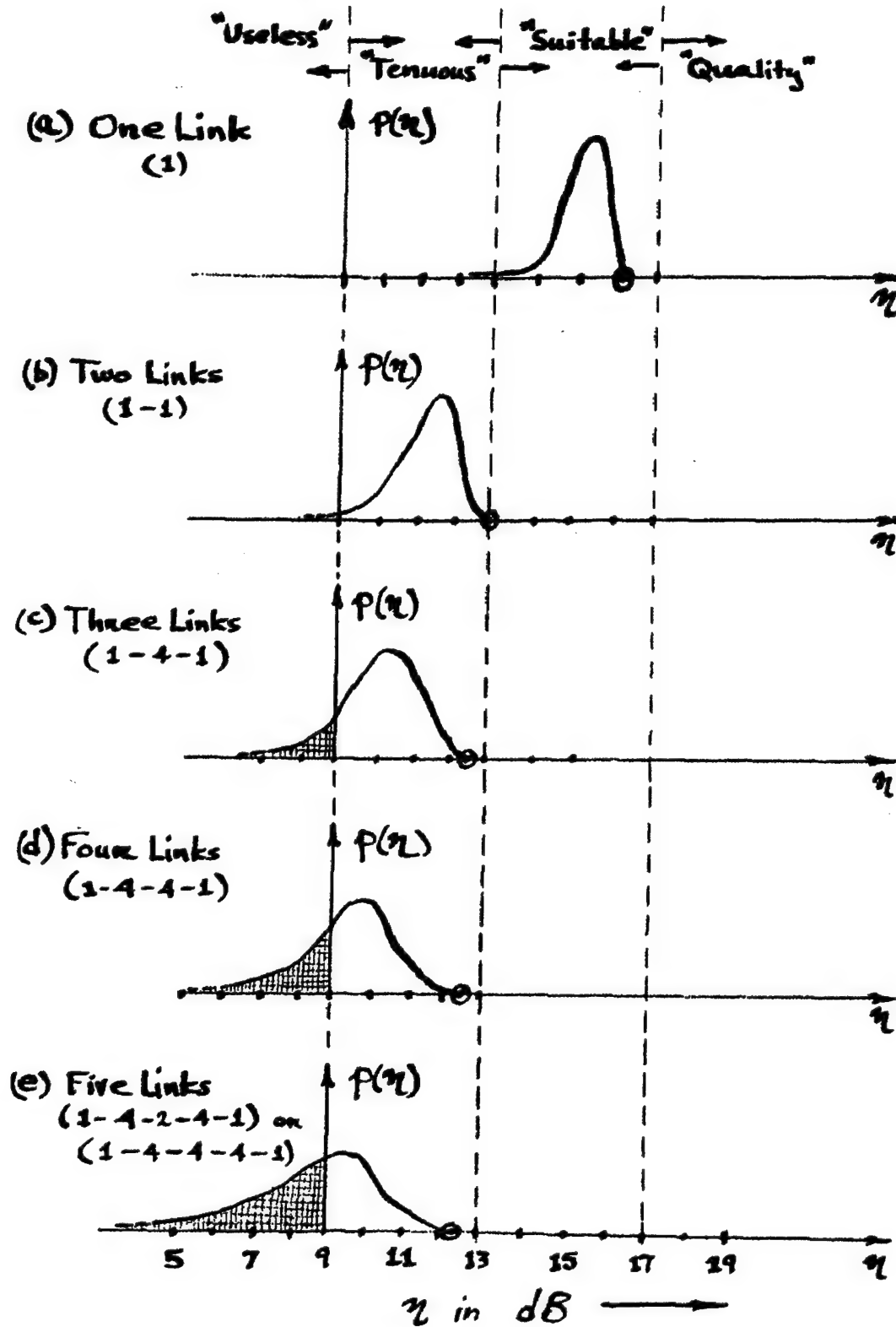
Figure 13 Illustration of the Probability Density
Function of the SNR Performance of Tandem Link Circuits

The variation in SNR performance is due to the channel error rate. Such a link remains "Suitable" for error rates as high as 1% (refer to Table 8). Note that such a link may become "Tenuous" or "Useless" as the error rate increases.

The three-link circuit of Part (c) displays a wider range of variability. The wider range is descriptive of such a circuit because the performance is vulnerable to additional impairments due to noise, offsets, and nonlinear effects. The area under the p.d.f. curve is unity, so the fraction of the area which lies to the left of the $\eta = 9$ dB line is the probability that the circuit performance will be "Useless." We note that as additional links are added to the circuit the p.d.f. curve becomes more broad and the probability of "Useless" performance increases.

The discussion of the results of Table 17 and Figure 13 leads to the following conclusion:

Conclusion #1. Pairs of users which are connected by a circuit that includes two links of CVSD-16 are potentially isolated. Such circuits should be avoided.

We note that such circuits may be avoided if digital transmission connections are provided between communities of CVSD-16 users. If so, direct one-link circuits are possible.

Selected examples of circuits with additional impairments were considered in Chapter 4. The discussion of Figure 12 demonstrates that user-pairs in the present ATACS system may be isolated due to level offset impairments. Links of PCM-48 are vulnerable to such impairment because of the twin-peak SNR versus ISL characteristic illustrated in Figure 10. The performance of circuits of tandem links in the transitional system of the 1980's will be affected.

Conclusion #2. The SNR performance of U.S. Army 48 kB/s PCM is sensitive to input signal level. This fact is a consideration which affects the versatility of employment of PCM-48 in tandem-link circuits in the transitional system of the 1980's.

The judgements in Chapter 4 of what tandem-link circuits are "likely" in the 1980's were based on an optimistic point of view. That is, an implicit assumption was made that the interoperability solutions (such as the "Dual-Rate" method) will be implemented within the U.S. Army system. If such solutions are implemented, then the system design for the 1980's is manageable and does not cause isolated user-pairs.

In contrast, suppose an analog mode of interface were implemented to connect CNR subscribers to the switched network. The impact of such a choice is that a circuit from a CNR user to any other CVSD-16 user (another CNR, and MST user, or a CVSD user in the DCS or Navy System) would include at least three links, two of which are CVSD-16. Thus, such user pairs would be potentially isolated. The point is that such a system design choice would guarantee that certain user-pairs would be isolated. [6]

Conclusion #3. The design for the U.S. Army tactical system of the 1980's is workable, and incorporates interoperability solutions which, if implemented, will avoid the occurrence of isolated communities of users.

The analysis of tandem-link circuit performance detailed in Chapter 4 is based upon a simplified model which incorporates SNR as a measure. Such analysis is useful to estimate performance, but many factors present in an actual working system are ignored in the simple model. Thus, further testing is necessary to determine what circuits of tandem links should be permitted in the system of the 1980's. The results of such testing will be helpful in the design of the switched system routing protocal, and will provide useful guidelines for system layout.

<u>Conclusion #4.</u> Further testing of the performance of tandem-link circuits should be conducted to determine the utility of connections which include combinations of CVSD-16, CVSD-32, PCM-48, and analog links.

As the transition progresses during the 1980's, digital switches, transmission, and user terminals are fielded. As time passes, an increasing fraction of users are equipped with digital terminals. Concurrently, the switched network becomes increasingly capable of providing direct (one-link) circuits to connect digital users. Thus, as the transition proceeds, an increasing fraction of user pairs are capable of establishing a direct circuit. (On such a direct path, the information signal remains in digital form from user to user.) Two observations follow; one relates to circuit performance, and the other to data communications capability.

From the discussion of Figure 13 it follows that the variability of the performance of voice circuits is decreased as the occurrence of tandeming is decreased. That is, circuit reliability is enhanced by minimizing the occurrence of tandeming.

We note that user-pairs that may communicate on a one-link circuit gain a direct digital path from user to user. Thus, such user pairs may pass data communications. This fact appears to represent an important communications resource. Early in the transition period, users that require direct data communications must be equipped with data modems. (which convert the data signal to "quasi-analog" form). The modem permits a 600 or 1200 B/s data path to be established on an analog circuit. The use of PCM-48 and CVSD links on such a path leads to uncertain performance of data modems. (For example, dedicated circuits may be required to gain satisfactory modem performance.) In contrast, note that user

pairs served by a direct link may convey data communications at 16 or 32 kB/s and no data modem is required. The contrast in capability is striking. One page of facsimile data, for example, would be conveyed 27 times more quickly on a 16 kB/s link than on a 600 B/s circuit. In addition, we note that the growth in data communications capability occurs without additional cost. The capability is inherent in the service which the system provides to digital users.

Conclusion #5. The transition of the 1980's should be planned and implemented to minimize the occurrence of tandeming. Such an approach accomplishes two positive results. First, the variability of voice performance is reduced, so the reliability of user service is enhanced. Second, to maximize the fraction of user-pairs that may communicate on a direct link results in a significant increase in data communications capability.

The DoD System

Let us broaden our perspective to consider the overall DoD system. The discussion of "interoperability between systems" which begins on page 66 includes a global point of view in which the overall system is considered to be the union of various regions (see Figure 9). Each backbone and access region of the system represents that transmission and switching network and community of users that are equipped with the "next-generation" of hardware. Thus, each backbone region is a homogeneous digital network that provides direct one-link circuits to backbone subscribers. As the transition proceeds in the 1980's, an increasing fraction of the total system is converted to the configuration of Figure 9.

The "Dual-Rate" method of interoperability is now being planned and implemented to operate at each of the system boundaries joining 16 kB/s and 32 kB/s regions. The method permits circuits which originate in, traverse across, or terminate in a 16 kB/s region to be established as

direct (one-link) circuits (at 16 kB/s). If such a method of interoperability is implemented then all CVSD subscribers of Regions I, II, III, V, and VI may communicate on a one-link circuit. Thus, voice performance is reliable, and direct data communications is possible.

Conclusion #6. The system designs of the various regions of the overall DoD system are converging toward a common target. The target system is a 16 kB/s digital transmission and switching network which provides one-link circuits between CVSD subscribers. Such a system contains no isolated subscribers. Voice performance is reliable and direct data communications is possible.

Some secure voice subscribers of the DoD system may not obtain 16 kB/s access, but must access the system over a narrowband analog channel. The channel rate on such channels is limited to 5 kB/s or less. Thus, a need exists to extend secure voice service to narrowband users.

A research and testing effort coordinated by the DoD Narrowband Digital Voice Consortium is focused on selecting a single narrowband digital voice terminal for use throughout the DoD system for those users who require narrowband service. The candidate terminals include channel vocoders, and LPC vocoders (among others). (We have adopted the label "LPC-2.4" to identify the narrowband terminal which will be selected.) The test results indicate that the two-link circuit composed of one link of LPC-2.4 and one link of CVSD-16 is acceptable if channel errors are not excessive.

To facilitate the discussion, let us assume that one link of LPC-2.4 degrades circuit performance in approximately the same measure as one link of CVSD-16. Thus, one link of LPC-2.4 is "Suitable." A two-link circuit of CVSD-16 and LPC-2.4 delivers performance which falls approximately at the "Tenuous/Suitable" boundary. Such a two-link circuit is useable, but has little margin to suffer additional impairments.

We conclude that LPC-2.4 and CVSD-16 users are not isolated from one another if a two-link circuit can be established (and if additional impairments are suitably small).

Conclusion #7. A viable architecture for DoD secure voice may be based on a 16 kB/s backbone system which provides direct-link circuits between CVSD subscribers and which provides narrowband access by LPC vocoder users. In such a system, circuits between similar subscribers (CVSD or LPC) are of one link, whereas circuits between dissimilar subscribers are two-link ones. Further testing should be accomplished to verify that the LPC vocoder to be fielded may operate successfully in tandem with CVSD-16.

We may describe such a system as a uniform 16 kB/s digital system with narrowband (LPC) access. Such a system displays inherent integrity because any secure voice user may communicate with any other user.

Conferencing

Conferencing is a user service that permits three or more subscribers to join in the same voice conversation. To provide such service the various circuit paths must be interconnected at one or more "bridge" junctions. The traditional method of conferencing is an "analog bridge." Such a bridge interconnects all conference circuits at a common point (on a two-wire analog basis).

The design of the TTC-39 switch incorporates an analog bridge for conferencing. [7] Such an approach adds one additional link to each conference circuit path. The analog bridge approach will permit satisfactory conference service among the CVSD-32 subscribers of the multichannel system. In addition, one CVSD-16 subscriber may be included in such a conference. But notice that if two or more CVSD-16 subscribers are included, then the interconnecting path is a circuit (of two or more links, two of which are CVSD-16) that is "Tenuous," and potentially not

useful. Further, if both CVSD-16 and LPC-2.4 users are included in a conference connection, then paths occur which include two or three links of LPC-2.4 and CVSD-16 ("Tenuous" or "Useless" paths).

Conclusion #8. A method of secure voice conferencing should be adopted which does not add an additional link in the circuit path. One such method is a push-to-talk (half-duplex with interrupt capability) approach with a four-wire "one-to-many" digital switching matrix as the conferencing bridge. [8]

Technical Observations

The results of Chapter 4 demonstrate that the irregular SNR versus ISL characteristic of PCM-48 is a factor which affects the performance of tandem-link circuits. CVSD includes adaptive slope companding, a technique which leads to a SNR versus ISL characteristic which is roughly "flat" over a range of ISL. But, how wide a range? The range is determined by the choice of the slope companding ratio (a design choice). The current design incorporates a ratio of 12 to 20. The use of a voice-actuated gain adjustment device (VOGAD) may further extend the dynamic range.

The point of this discussion is that the SNR versus ISL performance of CVSD is a key determinant in the performance of tandem-link circuits. The design of the CVSD implementations to be fielded in various equipments (e.g., the CNR, the DSVT, and the IMU of the TTC-39) may include differences of use of VOGAD, slope companding ratio, optimum ISL, and filtering. Such differences may affect both end-to-end performance between non-identical terminals and the performance of tandem link circuits. Improving the companding design may permit improved performance in circuits which include links of LPC and CVSD.

CVSD design improvements might be incorporated in the initial production version of some equipments, and retrofitted in others. Im-

proved algorithms that are not end-to-end compatible with CVSD should be avoided; otherwise an avoidable "tandeming boundary" would be created.

The variable channel rate strategy summarized on page 64 is a terminal-oriented approach in which all secure voice users would be equipped with a terminal capable of various channel rates (e.g., from 600 B/s to 16 kB/s). The idea is that the terminal could adapt to a mode of operation at a bit rate that may be accomplished on a particular circuit path. In stressed transmission or network conditions the terminal could select a low rate (600 or 1200 B/s). In unstressed conditions the higher rate would provide improved quality. Such users could operate on an end-to-end digital basis (i.e., on a one-link circuit).

There are several uncertainties and apparent disadvantages to the variable channel rate proposal. First, the complexity of such a terminal is greater than that of LPC. It seems unlikely that such a terminal may be produced in the early 1980's which will meet the difficult size, weight, and power constraints of the manpack, tactical user. (Terminal cost is a related issue.) Second, the approach has implications that reach far beyond the terminal design. Adaptive channel rate features would be required in all the transmission, switching and system control hardware to be fielded. These fundamental system design implications may not be incorporated in the TRI-TAC hardware now in development unless a significant delay is imposed.

On the other hand, the central idea of the variable rate strategy might be incorporated into the narrowband terminals to permit operation at several rates (e.g., 1200, 2400, and 4800 B/s), depending on the available channel rate. The signal format could be constructed in a very simple way, so that the 2400 B/s signal is a subset of the 4800 B/s one, etc.

Thus, improved performance might be gained on some circuits, including tandem links with CVSD-16.

Tactical Observations

The tactics, doctrine, and organization of the U.S. Army are evolving toward increased centralization of resource control. This trend generates new requirements for responsive, flexible teleprocessing and telecommunications.

The "new tactics" depends upon an integrated intelligence system to permit the commander to "read the battle" quickly, and upon a responsive command and control system to permit rapid concentration of forces at the decisive region of the battlefield. The doctrine to guide the joint coordination of combined arms to implement the "new tactics" is incomplete. But it is clear that responsive telecommunications is one necessary component of the evolving doctrine.

The Echelons Above Division (EAD) organizational structure imposes increased reliance on responsive long-haul telecommunications. Similarly, recent structure changes of logistics organizations (e.g., the Direct Supply System which emphasizes "throughput distribution") cause increased reliance on telecommunications.

The trend is clear. Responsive telecommunications needs may increase steadily over the next two decades. Secure voice and data communications resources are the key user capabilities needed to satisfy the potential new requirements.

The growth in digital communication needs (both secure voice and data) is related directly to the limitations of tandeming in the transitional communications system of the 1980's.

The uniform digital structure for secure voice provides direct (one-link) circuits between user pairs. Thus, the reliability of secure voice service is improved, and an additional data communications capability is available.

Notice that the uniform structure is not sensitive to detailed knowledge of the needline requirements. As new needlines are created, the system may satisfy the new requirements without a change in structure. The allocation of transmission link capacity may be adjusted periodically to match the actual traffic patterns.

The insensitivity to needlines is a dimension of flexibility of operational capability that appears to be very important in future combat operations. Our ability to define the needlines required in the 1980's and 1990's is modest. New weapons systems, tactics, doctrine, and organizational structures are certain to appear in the next two decades. We may only speculate today about what telecommunications needlines will become requirements in the 1980's and 1990's.

Acknowledging the ability of the uniform system to satisfy new needlines in a responsive manner, we observe that one figure of merit which describes the utility of the transitional communications system is the number of user-pairs that may accomplish one-link circuits.

Conclusion #9. The capacity of the transitional system of the 1980's to satisfy new needlines will be enhanced by maximizing the number of user pairs that may communicate on circuits of one link. (Notice that this conclusion is a restatement of conclusions #5 and 6.)

The interoperability solutions now in planning provide transparent interoperability at the various system boundaries in the overall DoD system. Such interoperability permits direct one-link circuits to traverse the boundaries. Thus, the occurrence of isolated user communities is

avoided. Cost and schedule trade-offs may be proposed which would implement a system configuration which does not incorporate the interoperability solution at one or more of the system boundaries. Such a decision would result in isolated communities of users and in decreased flexibility to satisfy new needlines.

Conclusion #10. Any system proposal that would not permit one-link circuits to traverse a system boundary should be carefully considered in terms of the communities of isolated users and the decreased ability to satisfy new needlines that would result.

Notice that the interoperability solutions are not merely technical details that should be resolved by engineers. Rather, the solutions implemented will directly determine the capability of the communications system to support combat operations.

Summary

In this subsection we present a two-page discussion of conclusions, a concise summary of conclusions, and a listing of the detailed conclusions of the study.

Discussion. The performance of circuits in the DoD telecommunications system of the 1980's is dominated by the influence of tandem links of digital voice. Both clear voice and secure voice performance is dependent on the nature of tandem links arranged in series to interconnect subscribers. The various links that may be traversed in the hybrid transitional system include analog links and digital links of PCM-64, PCM-48, CVSD-32, CVSD-16, and LPC-2.4.

Both CVSD-16 and LPC-2.4 deliver performance that allows little margin for additional impairments, so that such links offer little versatility in tandem connections. PCM-48 and CVSD-32 performance provides additional versatility in circuits of tandem links.

In general, circuit performance is inversely related to the number of tandem links connected in series. Thus, to minimize the occurrence of tandeming in the transitional system will maximize the performance of voice circuits.

The system transition plans of the Services and DoD Agencies are converging toward a digital objective system that will provide direct digital 16 kB/s links between subscribers. Such direct links avoid tandeming impairment and provide secure voice users the ability to pass data communications. Because the backbone system may be traversed without tandeming, the extension of secure voice circuits into the narrowband access regions of the DCS and the U. S. Navy may be accomplished on a two-link connection. If the two-link circuit of CVSD-16 and LPC-2.4 is satisfactory, then narrowband access subscribers may communicate with all CVSD-16 users served by the backbone system. Further, if "through-trunking" capability to convey 2.4 kB/s signals is provided by the backbone system, then LPC-2.4 access users could all be interconnected on a direct link. In such an objective system, satisfactory secure voice circuits may be established between every pair of users. For these reasons, the 1973 policy focus on 16 kB/s secure voice may result in DoD-wide interoperability in the future system.

The system plans of the U. S. Army, TRI-TAC, and the DCS incorporate the Dual-Rate method of interoperability at various system boundaries. The method is a workable one that permits direct digital connections to traverse the boundaries joining the (16 kB/s) MSA subsystem, the INTACS (32 kB/s) multichannel system of TRI-TAC assets, the (16 kB/s) U. S. Navy system, and the (16 kB/s) AUTOSEVOCOM of the DCS.

The U. S. Army system during the transition is a hybrid one in-

volving tandem links of PCM-48, CVSD-32, and CVSD-16. Pairs of subscribers
in the system will be isolated if an excessive number of tandem links is
required to establish a connecting circuit. The planning of the system
configuration during the transition should incorporate the goal that tan-
deming be minimized.

The performance of PCM-48 is sensitive to ISL. This fact is a
consideration which affects the versatility of employment of PCM-48 in
tandem-link circuits. Further testing is necessary to establish what
combinations of CVSD-16, CVSD-32, and PCM-48 will deliver satisfactory
tandem-link performance.

Concise Summary of Conclusions. Some isolated user pairs will
exist in the hybrid transitional system of the 1980's. The occurrence of
isolated user pairs may be minimized by providing digital transmission
paths to directly interconnect communities of 16 kB/s CVSD subscribers.

The system designs for the U. S. Army System and the overall DoD
System include workable interoperability solutions. By managing the tran-
sition in an enlightened manner, the reliabiality of secure voice communi-
cations may be enhanced and the operational capability to satisfy new data
communications needlines may be increased.

Detailed Conclusions. The detailed conclusions of the research
follow:

1. Pairs of users which are connected by a circuit that includes
two links of CVSD-16 are potentially isolated. Such circuits should be
avoided.

2. The SNR performance of U. S. Army 48 kB/s PCM is sensitive to
input signal level. This fact is a consideration which affects the versa-
tility of employment of PCM-48 in tandem-link circuits in the transitional

system of the 1980's.

3. The design for the U. S. Army tactical system of the 1980's is workable, and incorporates interoperability solutions which, if implemented, will avoid the occurrence of isolated communities of users.

4. Further testing of the performance of tandem-link circuits should be conducted to determine the utility of connections which include combinations of CVSD-16, CVSD-32, PCM-48, and analog links.

5. The transition of the 1980's should be planned and implemented to minimize the occurrence of tandeming. Such an approach accomplishes two positive results. First, the variability of voice performance is reduced, so the reliability of user service is enhanced. Second, to maximize the fraction of user-pairs that may communicate on a direct link results in a significant increase in data communications capability.

6. The system designs of the various regions of the overall DoD system are converging toward a common target. The target system is a 16 kB/s digital transmission and switching network which provides one-link circuits between CVSD subscribers. Such a system contains no isolated subscribers. Voice performance is reliable and direct data communications is possible.

7. A viable architecture for DoD secure voice may be based on a 16 kB/s backbone system which provides direct-link circuits between CVSD subscribers and which provides narrowband access by LPC vocoder users. In such a system, circuits between similar subscribers (CVSD or LPC) are of one link, whereas circuits between dissimilar subscribers are two-link ones. Further testing should be accomplished to verify that the LPC vocoder to be fielded may operate successfully in tandem with CVSD-16.

8. A method of secure voice conferencing should be adopted which does not add an additional link in the circuit path. One such method is

a push-to-talk (half-duplex with interrupt capability) approach with a four-wire "one-to-many" digital switching matrix as the conferencing bridge.

9. The capacity of the transitional system of the 1980's to satisfy new needlines will be enhanced by maximizing the number of user pairs that may communicate on circuits of one link.

10. Any system proposal that would not permit one-link circuits to traverse a system boundary should be carefully considered in terms of the communities of isolated users and the decreased ability to satisfy new needlines that would result.

RECOMMENDATIONS

The following concise recommendations are offered:

(1) The occurrence of tandeming of voice links should be minimized in the transitional system of the 1980's.

(2) Further testing should be accomplished to verify what tandem connections of PCM-48, CVSD-32, CVSD-16, and LPC-2.4 deliver satisfactory performance.

(3) Further testing should be accomplished to verify that the narrowband digital voice technique selected for use in DoD (e.g., an LPC vocoder) may operate successfully in tandem with CVSD-16.

(4) A method of secure voice conferencing should be adopted that does not add an additional link in the circuit path.

(5) An objective system architecture for secure voice in the DoD system should be adopted that is based upon a uniform 16 kB/s backbone network which provides direct one-link circuits to CVSD users and provides access to narrowband users.

SUMMARY

The research reported in this thesis examines some of the issues involved in the transition of DoD communications systems to digital operation. The focus of the study is the performance of digital voice links in series connections.

The transitional system of the 1980's will be a hybrid network in which circuits will be composed of a variety of analog trunks and a mixture of digital links which employ PCM-64, PCM-48, CVSD-32, CVSD-16, and LPC. Both CVSD-16 and LPC links provide little performance margin for additional impairments due to tandeming and noise. Thus, the system configuration must minimize the occurrence of tandem links.

Isolated pairs of subscribers will exist in the transitional system if an excessive number of tandem links is necessary to establish connecting circuits. The design now in planning for the U. S. Army System and for the overall DoD System is a coherent one which may avoid the occurrence of isolated user pairs. The overall system is converging toward a uniform 16 kB/s digital network which provides one-link circuits between CVSD subscribers. Such a uniform system offers reliable voice performance and direct data communications capability.

To minimize the occurrence of tandeming of voice links in the transitional system will maximize the reliability of secure voice performance.

CHAPTER END NOTES

1. These impairments are unpredictable to the extent that they may not be controlled by the communication system manager. For example, an analog junction link (at a "crossover" tandem point) that occurs within the TTC-39 switch is "well controlled" and should add .5 dB or less impairment to the circuit. At the other extreme, a wireline analog link in the series path may contribute 3 dB or more additional impairment due to level offset and noise. If the wireline trunk may be selected to complete the circuit path, then the impairment is beyond management.

2. Several issues of user acceptance of the communications service relate to this discussion. One may speculate that circuits of 13 dB SNR may gain user acceptance if such performance is reliable. In critical situations any useable circuit (however tenuous) may be a welcome resource.

3. These phenomena are physically distributed and are beyond orderly management. The phenomena include all sources of noise (thermal, atmospheric, radio frequency interference (RFI), intermodulation, jamming, power system hum, etc.), attenuation, and nonlinear effects on analog links. Digital link performance is a function of the error rate (and the distribution in time of the errors) --which is itself a function of noise, attenuation, and nonlinear effects.

4. For a discussion of the p.d.f., see George R. Cooper and Clare D. McGillem, Methods of Signal and System Analysis (New York: Holt, Rinehart, and Winston, Inc., 1967), p. 283. The area under the

curve is unity, i.e.,

$$\int_{-\infty}^{\infty} p(\eta)\, d\eta = 1 .$$

The probability that a circuit delivers SNR performance of at least η_1 dB is given by

$$P[\eta \geq \eta_1] = \int_{\eta_1}^{\infty} p(\eta)\, d\eta .$$

Thus, the fraction of the area under the curve to the right of the $\eta = 13$ dB line is the probability that the circuit is "Suitable" or better. (Similarly the fraction of the area to the left of $\eta = 13$ dB is the probability that the circuit is "Tenuous" or worse.)

5. An extensive program is undertaken periodically in the Bell System to measure the statistics of various circuit impairments. Such data is useful in estimating the relative performance of telephone service.

6. In a personal communication on May 18, 1976, Mr. Loren Diedrichson of the TRI-TAC Office confirmed that engineering implementation is in progress to accomplish the "Dual-Rate" method of interoperability between the tactical multichannel system (Region I of Figure 9) and the MSA subsystem (Region II), the U. S. Navy 16 kB/s subsystem (Region V), and the DCS (Region VI). Further, a digital mode of interface for CNR users (Region III) is recommended in the "External Interface" planning document now being distributed by the TRI-TAC Office.

7. In a personal communication on May 18, 1976 Mr. Richard J. Linhart of the Switching Branch, Equipment Division, Engineering Directorate of the TRI-TAC Office confirmed that an analog bridge is the current design choice for the TTC-39 switch.

8. Issues of cryptosynchronization associated with this approach

are beyond the scope of the discussion.

APPENDIX

APPENDIX I

TRI-TAC DEVELOPMENT AND ACQUISITION PROGRAMS

The Implementation Plan of the TRI-TAC Architecture details the strategy for transition in terms of hardware for the following categories:

(1) Switching,

(2) Static Subscriber Access,

(3) Mobile Subscriber Access,

(4) Trunk Transmission (Surface),

(5) Trunk Transmission (Space),

(6) Communications Control,

(7) External Interface, and

(8) Ancillary.

A concise summary of the objectives of the developments in each of the subsystems (1) through (7) is given in Table 1 through Table 7 on the following pages. The summaries are extracted from the Implementation Plan.

Table 4 contains two abbreviations not previously introduced; (1) Engineering Order Wire (EOW), and (2) Built-In Test Equipment (BITE).

Table 1
TRI-TAC Developments, Switching Subsystems

1. AN/TTC-39

This development is the heart of the Phase I architecture. It is
expected to use its hybrid nature and universality to bridge the
gap between the predominant analog inventory equipment and the dig-
ital devices now entering the inventory. It is full automatic,
under stored program processor control. The circuit switch is modu-
larly expandable from 300 to 2400 external lines. The message switch
has a 25 and 50 line configuration. The switches have the capabil-
ity for 100 percent security on digital loops and trunks.

2. COMSEC Module for AN/TTC-39

The COMSEC Module for the AN/TTC-39 will use electronic key distri-
bution, to attempt to eliminate the high logistics costs of hard
copy keying material. It will incorporate LSI techniques to pro-
vide reliable low powered security for digital loops and trunks for
voice, data and other record communications. It consists of Common
Equipment Facility racks with associated equipment, Trunk Encryption
Devices (TED), Loop Key Generators (LKG's) and AKDC. For the Static
Subscriber Subsystem Loop Group COMSEC a need exists to modify a TED
to permit "stand alone" operation. This effort is carried separately
as a Phase I Improvement program.

Table 2
TRI-TAC Developments, Static Suscriber Access Subsystem

1. Manpack Unit Level Circuit Switch

The switch will be designed to provide local service and se-
cure voice service for subscribers who are physically located
in areas served by current analog switching facilities. It
will be an all-digital 15-line/7 trunk portable circuit switch
designed to interoperate with the other digital unit level cir-
cuit switches and the AN/TTC-39.

2. AN/TTC-42 Unit Level Circuit Switches

The objective of this program is to develop 75 expandable to
150 line transportable circuit switch designed to interoperate
with the other unit level circuit switches, the AN/TTC-39 and
the analog circuit switches which are in the current inventory.

3. COMSEC Module for AN/TTC-42

The COMSEC Module for AN/TTC-42 must be operationally compat-
ible with the TENLEY equipment. Interfacing of inventory COMSEC
would be done at the AN/TTC-39 or CNCE and not burden the ULS.
It will provide the encryption and key distribution facilities
needed to secure the digital loops and trunks terminated at the
AN/TTC-42. The R&D effort will result in modification to the
DSVT and in a COMSEC version usable for the 30-line ULS of latter
phases.

4. Unit Level Store-and-Forward Modules

The modules should be able to be accessed on a switched circuit,
digital loop basis via either a unit level digital circuit switch
or an AN/TTC-39 to provide message switched service at the unit
level. It should provide capability for sizes of 12 and 24 lines.

5. Digital Subscriber Voice Terminal (DSVT)

This development will provide a digital telephone which will

Table 2 (Continued)

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interoperate with the digital side of the AN/TTC-39.   It pro-
vides the capability to terminate a data adapter to serve non-
voice terminal equipment and includes integrated encryption
equipment used to encrypt both signaling and traffic information.
Traffic information is encrypted on an end-to-end basis with the
key variable changed on a call-by-call basis. Provision will be
made to provide capability for use of a DSVT on an extension tele-
phone basis.

6. Modification of DSVT

The modified DVST will permit it to be used with the digital unit
level switches and the mobile subscriber access terminal facilities.
It will include provisions for operation in a net variable, oper-
ation in a 16 kbps push-to-talk mode, clear text ring signal and
a clear text traffic mode. It will reduce the amount of COMSEC
equipment which must be deployed with the digital level switches
as well as implement the 16 kbps half-duplex mobile subscriber
access subsystem.

7. Digital Non-Secure Voice Terminal (DNVT)

The DNVT is to be a telephone instrument optimized for non-secure
voice operation. It will interface directly with the TDMX of the
AN/TTC-39, the Digital Unit Level Switches and indirectly with the
DSVT via those switches. It will be provided with minimum essen-
tial service features to result in a low unit cost relative to
the DSVT.

8. Data Adapters

The Data Adapter will provide the full duplex digital interface
for data traffic dedicated into Store and Forward or circuit
switched with a DSVT, through the AN/TTC-39. Both synchronous
and asynchronous traffic will be handled. The complex version
will offer greater capabilities with respect to error control and
modes-of-operation.

9. Dedicated Loop Encryption Device (DLED)

The DLED will be a key generator which can be deployed in a dedicated

Table 2 (Continued)   140

mode to encrypt traffic carried on a loop. It will interoperate
with the LKG's in the circuit switch COMSEC modules, the DSVT's
deployed in conjunction with data adapters, and other DIED's. It
will operate at a variety of bit rates between 45 baud and 16/32
kbps.

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10. Tactical Digital Facsimile
(AN/UXC-4( ) (V))

This development will provide for high speed secure transmission
and reception of low resolution black, white and limited gray scale
graphic material over a wideband (16/32 Kb/s) digital channel. It
interfaces with the Data Adapter/DSVT and DIED. It also can oper-
ate over narrowband (3kHz) circuits interfacing with existing mo-
dems and COMSEC equipment. Operator selections will include data
rates of 2.4, 4.8, 9.6, 16.0 and 32.0 Kbps operation.

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11. Composition and Editing Device (COED)

The COED will provide assistance to users to assure that message
traffic is properly formatted and that header information is com-
plete. The COED assemblage will be used for composing, editing,
displaying, and monitoring message traffic. It will interface
directly with the Data Adapter/DSVT and DIED. Messages can be
manually entered in its internal memory from a keyboard, dis-
played for editing purposes, and then transmitted over a data
channel at rates up to 32 Kbps.

---

12. Loop Multiplexers

This family of loop multiplexers which will multiplex 4-wire,
full duplex, 32/16 kbps digital channels into serial bit inter-
laced groups and to demultiplex the groups into individual channels.
The family will include loop group multiplexers (LGM), remote loop
group multiplexers (RLGM) and remote multiplex combiners, (RMC).

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13. Stand-Alone TED

Modification of the existing Trunk Encryption Device (TED) to
permit the device to be deployed in a stand-alone mode and to
achieve reductions in size, weight and power due to the reduced

output requirements. It must provide bulk encryption for digital groups of 8/16 or 9/18 channels provided by a Remote Multiplexer Combiner or a digital Unit Level Switch.

14. Digital Loop Repeater

A digital loop repeater is required to be used either singly or in tandem to extend the transmission range of a single digital loop beyond 4 kms.

15. Shipboard UHF/LOS Transmission Facilities

This program involves a study and development of a multiple access/ discrete address transmission facility which could be employed for communications among ships traveling in a company and to provide network access for ships not equipped with SHF satellite transmission facilities. Transmission will be 16 kbps digital and secure subscriber terminal facilities will be designed to permit end-to-end secure operation with DSVT's located elsewhere in the TRI-TAC land-based or naval systems.

Table 3
TRI-TAC Developments, Mobile Subscriber Access Subsystem

1. <u>Basic Net Radio Interface Module</u>

This module will be used in conjunction with an analog switch to
provide single channel switched system access for secure net radio
users. It will include an operator control panel, a wire termina-
tion unit, a radio receiver/transmitter control unit which will be
equipped with up to four different interface control devices, and
a non-secure warning tone generator.

2. <u>Expanded Net Radio Interface Module</u>

This module will be used in conjunction with digital loop facili-
ties of the AN/TTC-39 or digital unit level switchboards to pro-
vide single channel switched access for secure net radio users. It
will include an operator-control panel, a modified DSTV, and a ra-
dio receiver/transmitter which can be equipped with up to five dif-
ferent interface control devices.

3. <u>Special Purpose Net Radio Interface Module</u>

This module can be used in conjunction with digital loop facilities
of the AN/TTC-39 or digital unit level switchboards to provide sin-
gle channel switched access, on a BLACK interface basis, for secure
net radio users. It will include an operator control panel, an auto-
mated radio receiver/transmitter control unit, a manual R/T override,
voice processing and crypto unit, and a wire line termination unit.

1. Tactical Digital Tropo, AN/GRC-XXX

This family of tactical tropo terminals will accommodate up to 72 channels (32 Kbps channel rate) and will operate in the 4.4 to 5.0 GHz band for ranges up to 200 miles. They will operate either in a LOS mode or in a Tropo mode. Secure digital EOW, BITE, interleaving equipment, and circuit quality monitoring capabilities will be incorporated. The three sets in the family, light, medium and heavy Tropo will provide between 1 Kw to 20 Kw output power. Set 1 (large) has 200 mile range, is the most capable family member and weighs less than 6500 lbs. Set 2 (medium) has 200 mile range, and weighs less than 5000 lbs. Set 3 (small) has LOS/Tropo range of 100 miles and weighs less than 3600 lbs.

2. Tropo Modem

The Tropo Modem must accommodate up to 32 channels (32 Kbps channel rate) to permit the AN/GRC-143 and AN/TRC-97 Tropo terminals to interoperate with each other and with the new AN/GRC-XXX Tropo terminals. The Modem will include the interleaving equipment needed to assure transmission performance which is consistent with that specified in the TRI-TAC error budget.

3. Modification of AN/TRC-97 Using AN/TCC-72 Components

The AN/TRC-97, which is in the Marine Corps and Air Force inventories, will be modified to improve near term interservice interoperability. The modification will accommodate the multiplexing, combining, and line encryption equipment of the Army PCM system (AN/TCC-72) to provide a 1152 Kbps digital transmission capability. It will provide for interoperability with the Army AN/GRC-143 and for trunking compatibility with the analog and digital matrix of the AN/TTC-39. The 1152 Kbps capacity will accommodate 24 channels PCM/TDM, 30 channels of mixed PCM/TDM and CVSD or 36 channels CVSD. It will also provide for pre-detection combining, increased power supply stability, improved receiver noise figure and operation with 60 Hz and 400 Hz prime power.

4. Modification of AN/GRC-103

Table 4 (Continued)

The existing radio assemblage will be modified to be compatible with the conditioned diphase interface with the Digital Group Multiplexers and the AN/TTC-39. The addition of provisions to facilitate remote reporting and equipment status information to an associated CNCE will also be provided. These modifications will provide for Phase I use of the inventory AN/GRC-103 radios where they are either collocated or separated up to 2 miles from the switch or associated CNCE. The modified assemblage will satisfy the 36 channel, low capacity (32 Kbps channel rate) digital trunking reuqirements of Phase I.

Table 5
TRI-TAC Developments, Trunk
Transmission (Space) Subsystem

1. TRI-TAC Small/Intermediate/Large Satellite Terminals

The effort consists of modifying existing satellite terminals to provide the characteristics and network flexibility called for in the system architecture. The modifications involve reconfiguration of assemblages and modification of modems, combiners/decombiners, and addition of buffers to accommodate a variety of digital trunk groups used in TRI-TAC and to appropriately interface with the synchronous TRI-TAC switching and control facilities. The system architecture calls for development of a small, intermediate (HUB), intermediate (SPOKE), intermediate (MESH), and a larger terminal for Phase I earth terminals. All except the large terminal will employ 8 ft. antennas.

2. TRI-TAC Control Terminal

This modification to current Army Satellite Tactical Control Terminals would permit them to be integrated into the TRI-TAC Communications Control architecture through provisions for expanded I/O interface equipment to interface operator personnel with the control subsystem. It will also provide for secure orderwire entry into each terminal.

3. Digital Combiner/Decombiner
   SPOKE/HUB

This equipment will handle 2, 3, or 4 AN/TTC-39 Trunk Groups and provide the flexibility needed to handle a mix of different groups required for optimizing system connectivity. In addition to providing the normal tactical satellite signal processing functions, it will perform the group buffering required for compensation of satellite movement. The combiner/decombiner will be integrated into the control communications terminal.

4. Orderwire Modem (Terminal Comm/Control)

This equipment will provide the functions associated with distributing satellite system control data between the communications terminals and the control terminal. It will be developed with the control

terminal orderwire modem. The orderwire modems developed will be integrated into the control communications terminals.

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5. Orderwire Security Device

It will likely be a modified available security device, tailored to fit the system orderwire concept. A device is needed to encrypt the orderwire transmission for both the control terminal and the satellite terminals themselves. The device will be integrated into the communications and control terminals.

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6. Universal Synchronous Multiplexer (USM)

This family of equipments will assure efficient usage of available satellite capacity at those locations where analog or mixed analog and digital traffic must be handled and access to an AN/TTC-39 or Unit Level Switchboard is not available. It will provide for conversion of 4 kHz analog channels to 32 Kbps digital channels and for multiplexing these channels into standard TRI-TAC digital groups of size 4-1/2, 9, or 18 channels. Analog/digital trunk groups in excess of 18 channels will be accommodated by combining the outputs of a number of USM's into a supergroup using a member of the DGM family. The USM will be integrated into the small terminal and selected CNCE's.

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7. SHF - Manual Trunk Channel Control Facilities

The use of FDMA during Phase I will require coordination of tactical accesses and frequency assignments, uplink power control, coordination with other non-tactical satellite accesses. Each control facility and each terminal operator will be able to initiate and coordinate the set-up of any temporary or extended period point-to-point and net radio circuit.

Table 6
TRI-TAC Developments, Communications Control Subsystem   147

1. Communications Nodal Control Element
   (CNCE) (AN/TSQ-111 ( ) (V))

   The CNCE will provide the processing capabilities to perform nodal
   functions. It will provide group patching, monitoring and test ca-
   pabilities, VDU/Keyboard/printer I/O and display devices. It must
   provide for interfaces with external systems and transitional equip-
   ment. The CNCE configurations will include existing equipment as
   well as equipment that is being developed under the TRI-TAC Program.
   Each CNCE will be implemented and equipped modularly to respond to
   the specific nodal environments in Phase I (e.g., expansion change
   in mix ratio). Type I is predominantly analog, processor equipped,
   two shelters, one of which is a management shelter. Type II is pre-
   dominantly digital, processor equipped, two shelters, one of which
   is management shelter, Type III - Equal amounts of analog and digi-
   tal, processor equipped, one shelter, non-expandable. Type IV - All
   digital, no internal processor but able to access remote processor,
   technical control capability, one S-250 type shelter. The CNCE, in
   conjunction with the CSCE and CSPE, will provide increased control
   capabilities and a near-real-time technical management capability.
   The CNCE will provide management and technical nodal control facil-
   ities to assign, monitor, control and manage resources and provide
   the interface between transmission facilities and users of the sys-
   tem.

2. Trunk Group Multiplex (TGM)

   The TGM will synchronously combine/decombine 2, 3 or 4 group inputs
   into and from a supergroup of maximum bit rate 4096 Kbps. The group
   inputs will be of the same family of group bit rates. It will be
   used in conjunction with the LGM, MGM, TED and TD-660 multiplexer.
   The TGM will be used in both shelters and vans and will be mounted
   in an equipment case.

3. Master Group Multiplexer (MGM)

   The MGM will combine/decombine up to 12 synchronous or asynchronous
   groups or supergroups. It will be deployed in conjunction with the
   CNCE AN/TSQ-111 and with remote ratio extension facilities. It will
   accept bit rates from 0.072 to 4.9152 mbps with output master group
   bit rates of 9.360 and 18.720 mbps. The MGM will have provision for
   two orderwire channels, each capable of operating at 16 or 32 kbps
   (switchable). The MGM will be used in conjunction with the LGM, TGM,
   and TED.

Table 6 (Continued)   148

4. Communications System Control Element
   (CSCE) (AN/TYQ-16 ( ) (V))

   Each CSCE will consist of personnel, information processing capa-
   bilities, displays and associated facilities to perform assigned
   monitoring and control functions. CSCE will be able to interface
   with the AN/TTC-39 on a limited automatic basis, and other invent-
   ory switches (e.g. TTC-30) on a manual basis. This program together
   with the CNCE and CSPE elements will provide a total integrated con-
   cept for near real time management functions. The CSCE will provide
   real time monitoring and data base maintenance of communications
   network status and near real time control over the allocation and
   use of resources within a portion of a deployed tactical network.
   Advanced development models of processor equipped CSCEs will be
   developed in Phase I, which will supplement current manual control
   functions.

5. Communications System Planning Element (CSPE)

   The CSPE will provide support for maintaining a file of available
   resources, collecting, storing and updating communications require-
   ments, issuing taskings, monitoring progress in implementation and
   testing and monitoring overall system performance. The other CSPE
   functions will be accomplished through normal staff action, during
   Phase I. The processor at the CSCE will be used for information
   storage and retrieval.

6. Short Range Wideband Radio

   This LOS multichannel radio will be used for various intrabase and
   down-the-hill employments serving as a network extension facility
   replacing the requirement for extensive cable service. This will
   be accomplished through modifications to the existing AN/GRC-199
   radio in the AN/GSQ-119 assemblage and the AN/GRC-144 in the AN/-
   TRC-138 assemblage. The modified assemblages will provide full
   duplex, multichannel links with full electric compatibility in the
   14.4-15 GHz frequency range. They provide capacity for digital
   traffic up to 20 Mbps over paths up to 8 Km, and 10 Mbps over paths
   up to 24 Km.

Table 7
TRI-TAC Developments, External
Interface Subsystem

1. <u>AUTOSEVOCOM/ULSB Interface Facility</u>

The facility will consist of a DSVT and necessary controls which
will provide a 32 Kbps operator controlled DCS interface with loops
from either a AN/TTC-39 or a ULSB. This will provide DSVT subscri-
ber direct access through the ULSB on a manual basis, through an
AUTOSEVOCOM switch where this facility would be located.

2. <u>NICS/NATO Interface Assemblage</u>

An interface shelter assemblage is required for the NICS/NATO in-
terface. The NICS/NATO Interface is expected to be automatic,
circuit switched and employing the Standard NATO analog electrical
and signaling interface parameters. A modified assemblage is re-
quired for use when the NATO interface point is not collocated with
the US tactical gateway switch. It will provide the conversion of
the standard NATO interface control signals into corresponding US
control signals which would then be passed on via the US interface
radio transmission link to the AN/TTC-39.

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